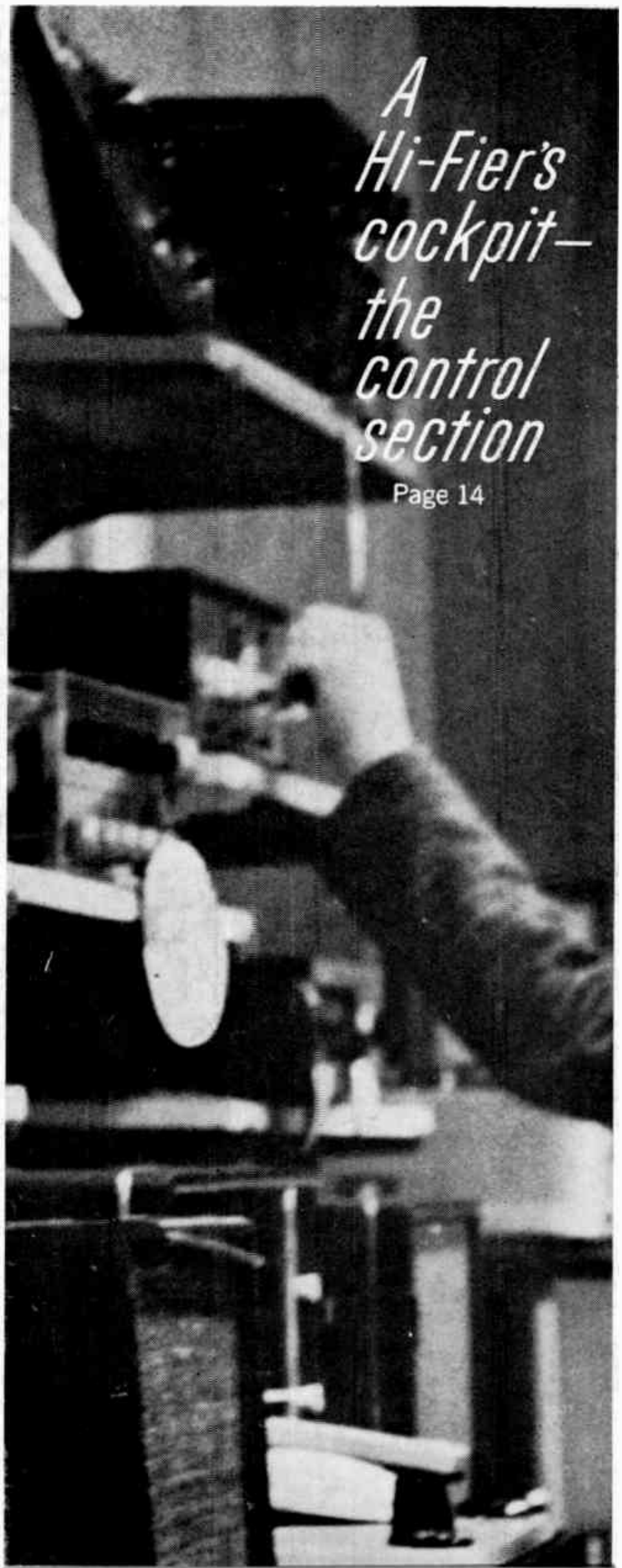
Audiofan

THE MAGAZINE FOR THE HI-FI ENTHUSIAST



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FEBRUARY 1966 VOL. 2 NO. 2 FIFTY CENTS

Amazing tower test proves superiority of Scott $300 receiver!

Revolutionary new "Field-Effect" circuits end cross-modulation problem <u>without</u> sacrificing sensitivity

Paul Hurd, Engineer-in-Charge of WHDH-FM, Boston, Mass., is shown checking new Scott 342 FM Stereo Receiver for sensitivity and cross modulation rejection. The 1250-foot FM stereo and television transmitting tower operates at multi-kilowatt power 24 hours a day.

The toughest place to test a solid-state FM tuner is *right* at a strong transmitter site. Being this close to the overpowering signal of the station causes ordinary tuners to "cross-modulate." A powerful station will appear at many points on the dial, obliterating other FM signals listeners want to receive.

To prove the superior cross modulation rejection of Scott's new Field Effect circuitry, the 342 FM Stereo Receiver was tested *right* at Boston's WHDH-FM transmitter tower. Here the radiated energy from the multi-kilowatt transmitter is at maximum level, and any susceptibility of a receiver to cross-modulation would be drastically evident. Not only did the Scott 342 reject cross modulation exceptionally well, but, equipped only with the normal FM dipole antenna supplied with the unit, the 342 picked up *31 stations* loud and clear in spite of impossible reception conditions.

Until development of Field-Effect circuitry by Scott engineers, it was impossible for an all-solid-state FM receiver to provide the listener with *both* high sensitivity *and* freedom from annoying cross modulation. This test strikingly demonstrates achievement of *both* desired results. Cross modulation rejection is at least 20 db better than conventional designs . . . and there is *no* sacrifice of sensitivity. In separate tests reported by Texas Instruments, Inc., a new Scott field effect transistor (FET) front end gave 1.6 microvolts sensitivity, over 96 db cross modulation rejection . . . hailed as an outstanding engineering accomplishment.

Scott . . . where innovation is a tradition

SCOTT®

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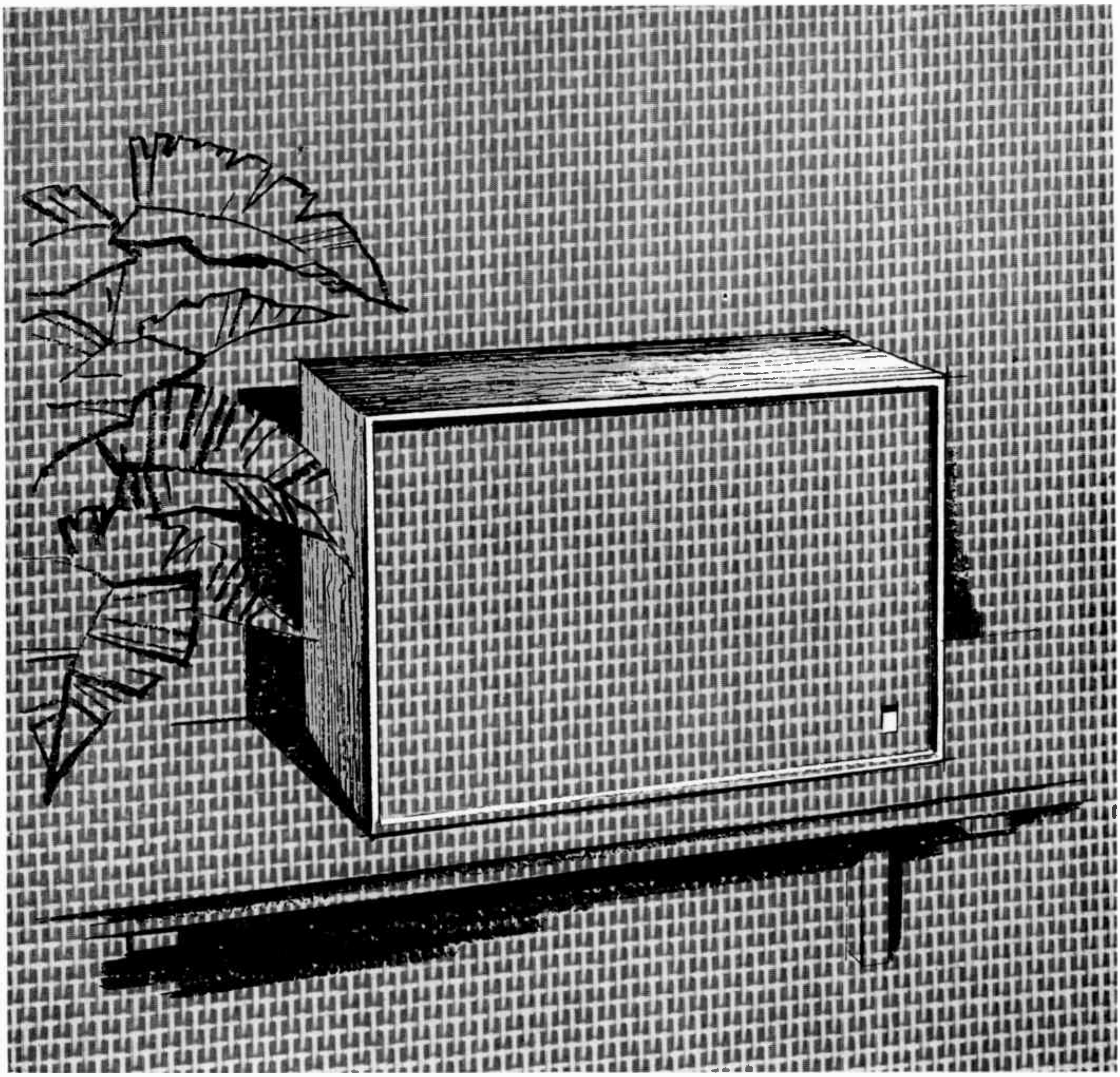


EDITORIAL

the hi-fi name game

It takes time and effort to absorb the meaning of words peculiar to a science, art or technology. It all comes a bit harder, you doubtlessly know, when such terms are related to a special interest; a hobby, if you will.

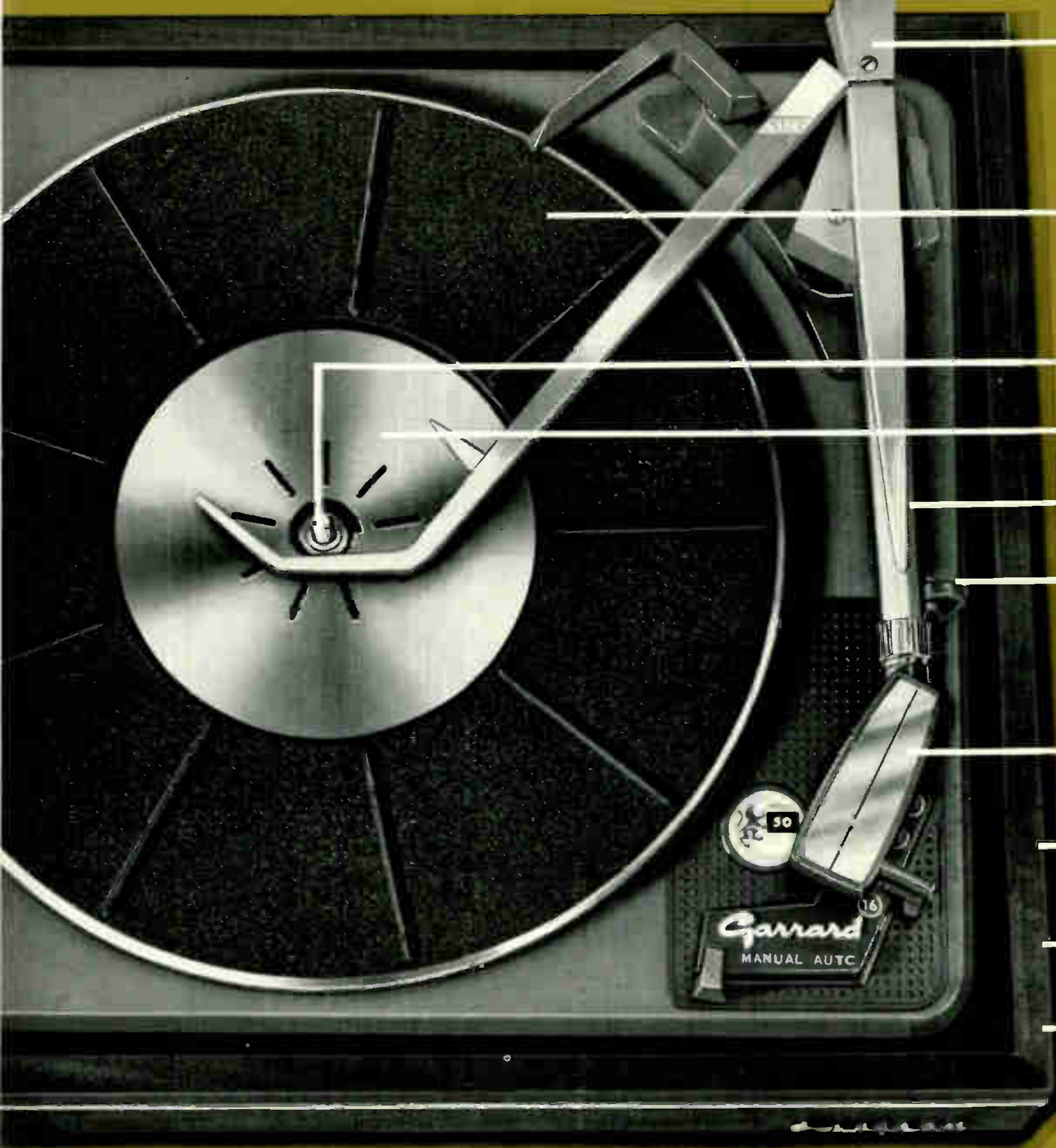
After all, it's not as if a school master pounded strange terms into your brain day after day. You picked up the "inside" language of hi-fi by yourself, as a labor of love, by reading magazines and books dedicated to high fidelity equipment, by joining an audio club, spending hours at a hi-fi show, obtaining advice on what to buy from audio specialists, and by exchanging ideas and information with friends.

New terms are introduced from time to time; old ones die. So audiofans have to keep on their toes to be abreast of changes. You might have observed such a change last month in AUDIOFAN, for example, when the abbreviation cps (cycles per second) was retired in favor of the new unit of frequency, "Hertz," adopted first by the International Committee on Weights and Measures and by the Standards Committee of the Institute of Electrical and Electronic Engineers (IEEE) only a few months ago.

In using the term "Hertz," which honors the German physicist who introduced the concept of an electric wave, Heinrich Hertz, we dispense with an anomaly: the phrase "cycle" is widely used to signify cycle *per second*, as in kilocycles. This is incorrect, of course. But we're now technically correct with the term, kiloHertz (kHz), where Hertz means cycles per second.

Isn't change wonderful?

THE EDITORS



Graceful cast aluminum tone arm is counterbalanced—first time this type of arm has been available in a popular priced unit. This feature alone gives the Model 50 particular significance—an automatic in the economy field which can track high quality cartridges for finer sound reproduction.

Oversized turntable with handsome mat is reminiscent of previous Garrard models in a considerably higher price echelon.

Two spindles—a convenient short spindle for playing single records manually; an interchangeable center drop spindle for automatic play when desired. Spindles remove for safety and convenience when taking records off the turntable.

In automatic position, Model 50 intermixes records of any size or sequence.

Stylus pressure adjusted with simple, accessible finger touch device, for correct tracking force, according to the cartridge manufacturers' specifications.

Super sensitive trip, with Dupont Delrin® to offset friction, operates with any high compliance pickup at correct minimal tracking force.

Shell is lightweight cut away type with extended finger lift for safety in handling. It plugs in . . . accommodates your widest personal choice of cartridges . . . can be removed from the arm instantly to change cartridge or service stylus.

Garrard 4-pole shaded "Induction Surge" motor, with dynamically balanced rotor, shielded from hum. Constant speed assured, free from vibration.

Ultra-compact—fits easily into any record changer space. Only 14⅞" left to right, 12½" front to rear, 4⅝" above and 2⅞" below motor board.

Simple installation. Fully wired for stereo, with a 4-pin, 5-wire system utilizing separate connection for ground, to eliminate hum. Leads connect to the changer with a built in Amplok plug (for AC) and a female twin phono socket mounted on the unit plate (for audio). Simply plug in at the player!

Important reading: 32-page Comparator Guide detailing all Garrard models. Write for complimentary copy to Garrard, Dept. GC-2096, Westbury, New York 11591.

This is the Model 50 *Garrard's most compact manual/automatic turntable. Despite its modest price of only $44.50, dealers large and small, in every part of the country, think enough of this unit to include it in the overwhelming majority of advertised systems which they pre-select. The dealer knows he can combine the Model 50 with the finest, most expensive brand name amplifiers, receivers, and speakers, and offer them to his most discriminating customers assured that it will be compatible and an enduring credit to his reputation. The dealer's recommendation is important to you. It is every bit as significant as the impressive list of features on this page, which the Model 50 incorporates. This is the lowest priced Garrard automatic . . . but all Garrards meet exactly the same strict standards of quality. Therefore, you can buy a Model 50 with complete assurance, and you will use it with pride as well as pleasure, for years to come.*

Garrard®

WORLD'S FINEST

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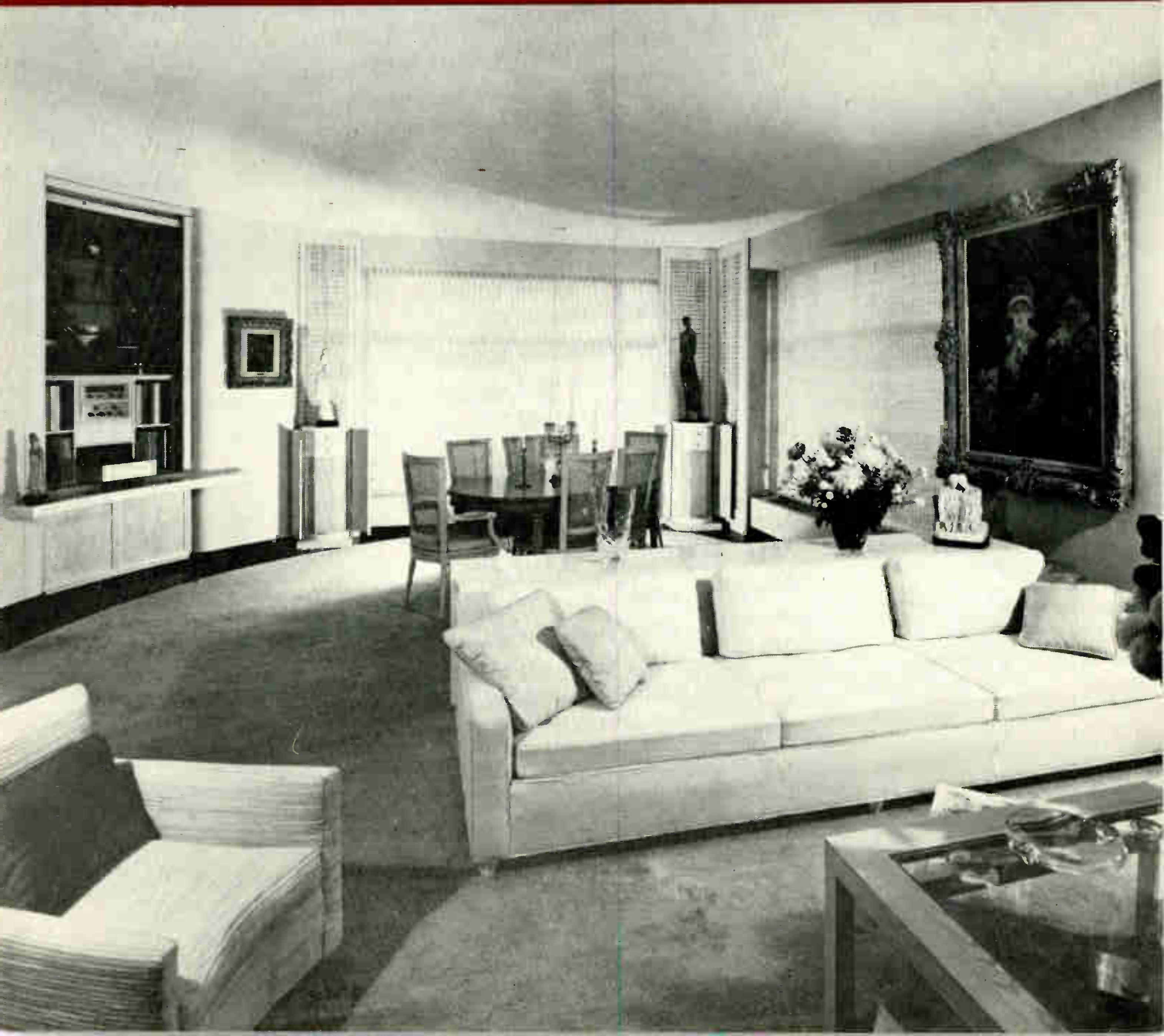
Room acoustics are largely influenced by room size and type of furnishings.



HOW GOOD IS YOUR ROOM FOR Hi-Fi?

here are tips on how to judge room acoustics

by Art Salsberg

Just sing in the bathroom shower and you'll quickly appreciate how a room's acoustic properties can influence sound! In this instance, your warbling hangs on around your ears, creating the familiar "bathroom" sound that disguises your voice (it might sound great to you, but, brother, ask someone else . . .). This phenomenon occurs because the room is so small and it's undamped by cushions, drapes, rugs and other sound absorbing materials.

Similarly your hi-fi listening room, be it a living room, den or basement, also affects sound reaching your ears. It "colors" sound. This is the reason why a speaker system sounds different from one room to another room.

The room might make our elegant super hi-fi stereo system sound mushy. On the other hand, it may mold the sound produced by your speakers in such a way that it's sharp to the point of irritation. Perhaps low bass is strangely absent, even though you *know* your equipment is able to go way down in frequency. Or maybe your room suffers from echos, or buzzing sounds at times.

But don't dispair, fellow audiofans. There *is* something you can do about it!

One of the keys to a room's "tone" is its reverberation time—how long sound persists. A sound doesn't stop instantly, you know. It dies away slowly, perhaps in ½ second, ¾ second, 1 second or more. When sound waves are produced by loudspeakers, the sound bounces from wall to wall and to other surfaces until the energy is almost completely absorbed. At this time the sound can no longer be heard. (This is taken to be one-millionth of its initial value or 60 decibels.)

It's obvious, then, that sound persistence or reverberation can be controlled to a great extent. Simply select the right materials. But this is easier said than done, for we're dealing with absorbing powers that differ from material to material, some of which we have no control over, the dimensions of a particular room, placement of furniture, and interference waves, among other factors. Let's tackle these considerations one at a time, so that at the end you can juggle them properly to control the sound reception in your room.

The final goal is to achieve the optimum reverberation time for your room. This might be ¾ second or so. If your room is too live, say with a reverberation time of 1 second, then you need more absorbing surfaces and, therefore, less reflecting surfaces. If the room is too dead, say ½ second, the converse is required: less absorbing surfaces and more reflecting surfaces.

An excellent sound absorber is an open window. It doesn't return energy that reaches it. (But think twice about using an open window to deaden a live room—your neighbors may pull out your fuse!) As the perfect absorber (it isn't really), its absorption coefficient is 1.00 per sq. ft. Felt is another good absorber, with a coefficient of 0.70, which means that a square foot of felt is equal to 0.70 square feet of open window space. Glass, on the other hand, with an absorption coefficient of 0.03 is a poor sound absorber, but a good reflector as are other materials tabulated in Table I.

how to determine reverberation time

You can actually get a rough idea of reverberation time in your room by using a table of absorption coefficients and a formula devised by an acoustics pioneer, Wallace Clement Sabine (1869-1919), a college physics professor:

$$\text{Reverberation Time} = \frac{0.05 \times \text{Room Volume in Cu. Ft.}}{\text{Absorbing Power}}$$

The absorbing power is calculated by multiplying the absorption coefficient of a material by its area in square feet.

This isn't as difficult as it seems. For example, let us assume your hi-fi system is located in a living room that measures 20' x 10' x 10'. The two long walls are constructed of plaster over brick; one short wall is covered by drapes; the second short wall is wood paneled less a 4 foot wide door; the ceiling is made of plaster on lath; the floor is fully carpeted; furniture consists of a long pillow-back couch, 2 wood chairs upon which are seated 2 people and a wood table.

To calculate absorbing power, simply multiply each material's absorption coefficient by its square foot area. Then simply add up all the absorbing powers. (The approximate absorbing power of furniture and people are already given.) In this hypothetical case, the following would be figured:

Plaster Wall	200 sq. ft. x 0.01	=	2.0
Plaster Wall	200 sq. ft. x 0.01	=	2.0
Panel Wall & Door	100 sq. ft. x 0.08	=	8.0
Draped Wall	100 sq. ft. x 0.20	=	20.0
Ceiling, plaster	200 sq. ft. x 0.03	=	6.0
Table		=	3.4
Couch		=	11.0
People (2) on chairs (2)		=	3.4
Carpeting	200 sq. ft. x 0.20	=	40.0
			95.8

Thus, the calculations are simple arithmetic when you get right down to it:

$$T = \frac{0.05 \times 2000 \text{ cu. ft.}}{95.8}$$

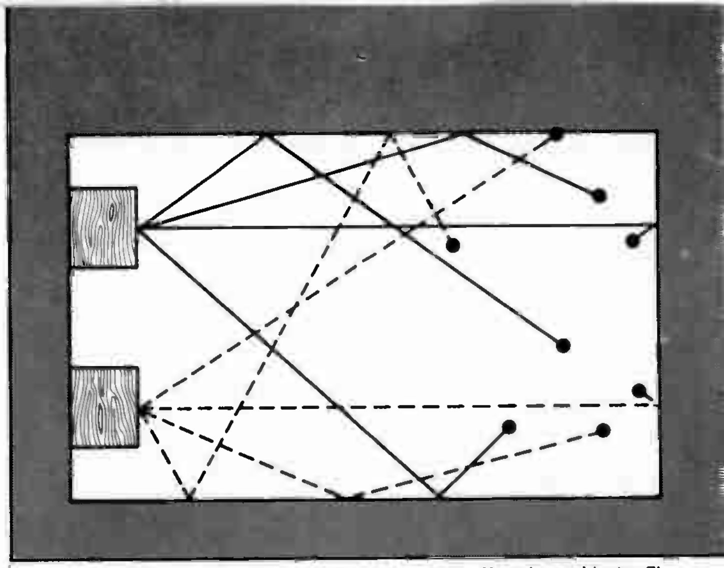
Reverberation time is therefore about 1 second.

This is a very approximate figure, to be sure. But that's all we really need. If other furnishings normal to a home were considered—a credenza, end tables, cocktail tables, lamps, pictures, and the hi-fi system itself—the reverberation time would actually be lower, perhaps 8/10 to 9/10 of a second, which would be about right for low frequencies in the vicinity of 125 Hz.

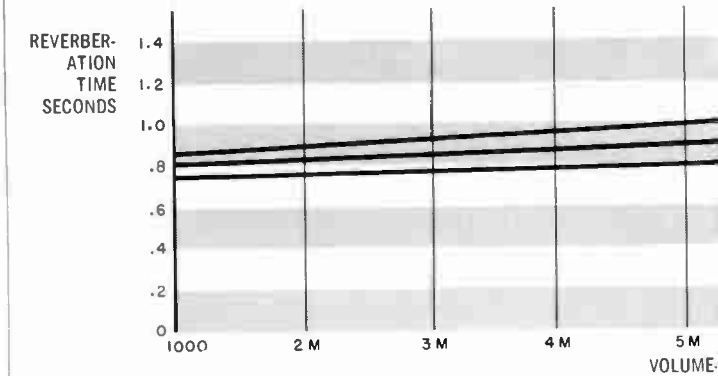
The absorption coefficients in Table I are given for low frequencies. They increase with rising frequency; a carpet's coefficient would be about 0.4 at 1000 Hz whereas it is about 0.2 at 125 Hz, for instance. Consequently, reverberation time is always less for high frequencies. This means that the room is not as live for high frequencies. With the stereo effect very dependent upon higher frequencies, it is therefore important not to have a room that's too dead.

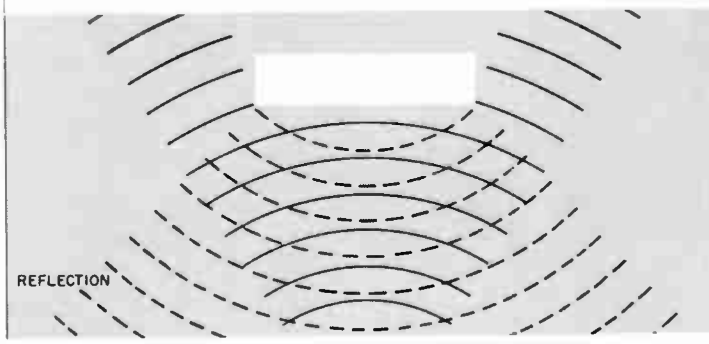
If a room hasn't enough reverberation time you could lose the all-important high frequencies, destroy the realism of middle frequencies and, if too dead, get dull thuds from vibrant low bass notes.

On the other side of the coin, a room with too many reflections, not enough absorbing material, can give rise to some distressing sonic problems such as echos (which might give the impression of poor transient response from your system), standing waves (where a reflected wave meets an incoming wave and causes a dead spot), and



Sound waves "sprayed" into a room bounce off various objects. Thus, at a given moment, the sound can be heard at many locations in a room. However, the sound may not be the same in each spot because



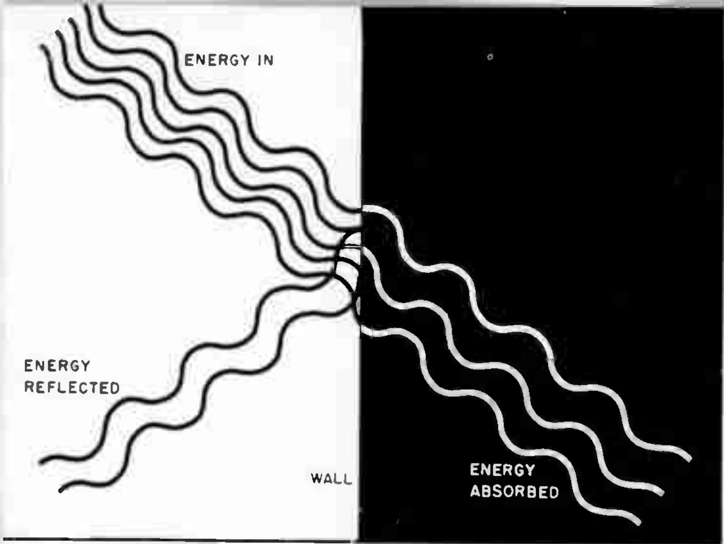


Using the above chart, you can quickly determine your room's "ideal" reverberation time. (Keep in mind, though, that other factors, such as standing waves, influence room acoustics.) At the right is the hypothetical room used as an example in the text.

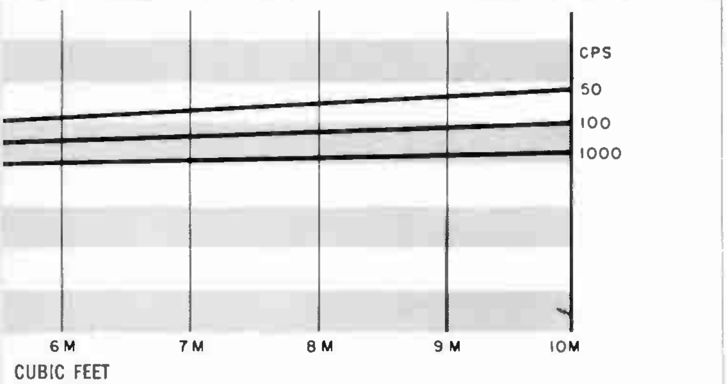
TABLE I

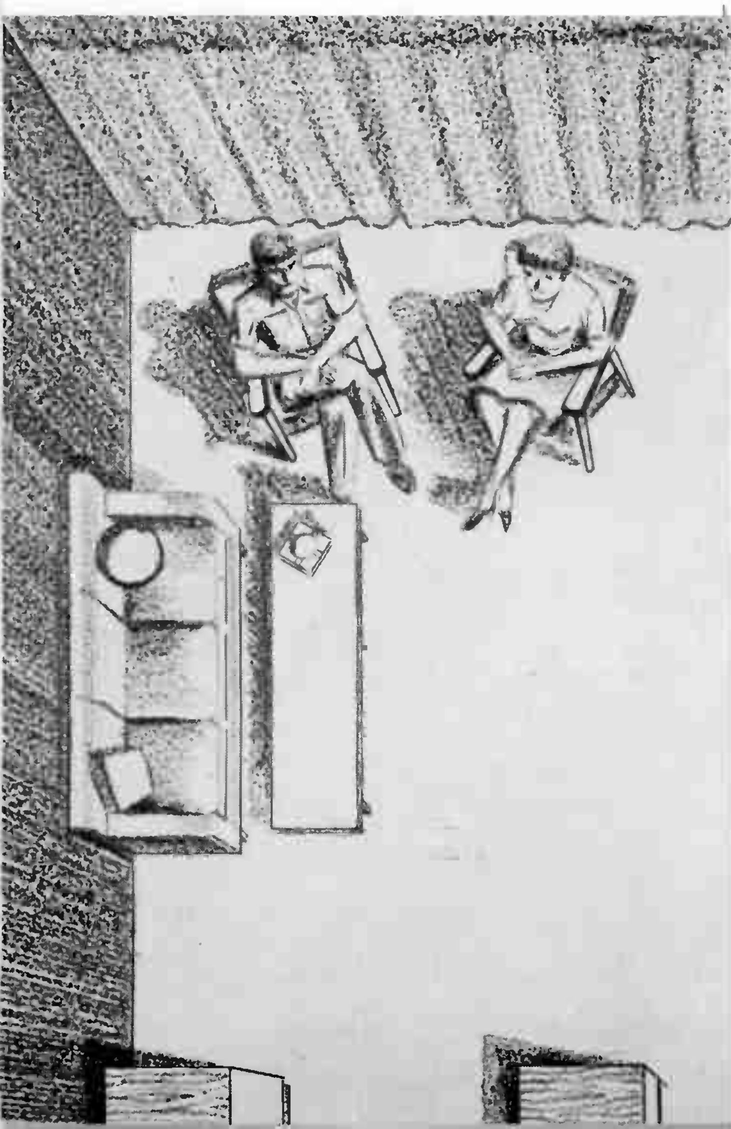
Absorption Coefficients (low frequencies)

Open Window	1.00
Brick Wall	.03
Plaster on Brick	.01
Mirror, Glass	.03
Wood Panels	.08
Plaster on Lath	.30
Wood Floor	.09
Carpet	.20
Linoleum on Floor	.05
Drapes	.20
Couch, each	11.00
Easy Chair, each	3.50
Auditorium Chair, each	0.30
Adults, each	2.00



different objects exhibit different abilities to reflect and absorb sound energy. As a result, certain frequencies might sound louder or softer in one or another location, even disappear completely.





a sense of too much brightness.

If our imaginary listening room didn't have wall-to-wall carpeting, it is interesting to observe, almost ½ second would be added to reverberation time (absorbing power would be decreased by 22 Sabins: 73.8 instead of 95.8). This would make the room decidedly too lively for good hi-fi sound.

So you see, furnishings do have a definite influence over your sound quality.

A longer reverberation time is required in larger rooms, as indicated in an accompanying chart here.

Of high importance, also, in achieving good sound reproduction is the size and shape of your room. The worst listening room you could have is a cube-shaped room. The resonances produced by a cubicle room would be unbearable to any true hi-fier. For example, a 12′ x 12′ x 12′ room would tend to have severe resonances at 47 Hz because 12 feet is one-half the wave length of that frequency.

An ideally shaped room for hi-fi would give sound resonances a chance to spread out. The ideal proportions of the average room for reproducing recorded or broadcasted music is said to be: Height, 1; Width, 1.6; Length, 2.5. Using dimensions, this would be Height, 8′ x Width, 12′9″, x Length, 20′.

What can you do about a room that isn't acoustically right? There are many ways to beat the beast. If your room is too live, you might consider drapes and/or carpeting. If you can't do this, why not consider a room divider screen or small scatter rugs, a large bookshelf filled with books, throw pillows, pictures on the wall or any other material to absorb sound waves?

If your problem is standing waves—you lose certain frequencies in specific spots in the room—try relocating your speakers. A corner location generally helps overcome this problem. You might also consider moving an easy chair or other large object to change the reflective pattern of the room. Placing a large object against the wall opposite the speakers will often eliminate standing waves, too.

Now what if your problem is a dead room? This presents some decided difficulties, unless you're prepared to dismantle your drapery, remove carpeting or change heavily cushioned furniture to light furniture. None of this is likely if the lady of the house has her way. Chances are, however, that you can liven the room for sound in other ways. For example, open the drapes to expose windows—glass is a great reflector—when you're playing your system. Perhaps you could convince your wife to have the drape lining removed. Then, too, why not consider pictures protected by glass, a large mirror, a metal cigarette stand, etc.

All these efforts are worthwhile, so long as you don't destroy the esthetic effects of your home. By adjusting your room for sound, you'll enjoy the full capabilities of your hi-fi system.

PROFILE OF AN AUDIOFAN

"if you listen to my tape collection 24 hours a day, it would take a month to hear it all," says medical doctor who doubles as an audiofan's audiofan

Four electrostatic speakers and two more conventional speaker systems dominate one wall of the 18' x 28' listening room.

Seated at his test equipment bench, with a fellow audiofan watching him, Dr. Close mounts a new phono cartridge in a pickup shell. The doctor uses the impressive array of test instruments to check out his recording equipment before going out to capture a live concert on tape.

The cabinet which houses our audiofan's hi-fi equipment was built to his specifications. Pull-out drawers, with equipment faceplates up, makes it convenient to adjust his tuner and preamp. From the extensive tape library seen in the background, it's evident that he's an avid tape recordist.

A Dr. Clarence C. Close, of Northbrook, Illinois, just north of Chicago, takes audio equipment seriously. He applies the disciplines of his profession to his listening, going at it with the scientific approach of a doctor, the precision of a technician. For to him there's a challenge in capturing for his hi-fi room near-perfect reproductions of sound—from the night sounds of insects in a field to the roll of thunder and the magnificient harmonies of an organ or the whispered hush of a percussionist's hand, silencing the timpani's reverberations.

"I'm a purist," he says, as he describes the stereo high fidelity system he's built over the years to form what he now considers his "ultimate system."

The system isn't "complicated unnecessarily with remote controls or remote speakers," but has the simplest and most direct circuitry, since he feels the signal may be degraded when it is sent through pads or networks. "And besides, there are far less problems in tracing any difficulties," Close explains. "The only place this system is played is in this room. It's not used for background music. I have other means for that. I think of it as a research tool, but I still get a great measure of pleasure out of just listening to it," he added.

The ultimate system, housed in a room designed for it, includes a Marantz model 7 console and Marantz FM tuner; four Marantz model 9 70 watt amplifiers, with one KLH 9 speaker for each of them, and two KLH 1 dual woofers to reinforce the bass end when necessary; two Ampex 354 stereo tape sources; a Fairchild model 412 turntable; a Shure Solophone and Sharpe headphones; a Shure SME 12 inch tone arm, and a veritable stable of cartridges because "certain of them work better for certain kinds of records." Among the cartridges are the Ortofon SPE-T, the Shure V-15, a Grado Laboratory model, and a Decca Mark II.

(A second system, alluded to above, is located in a bedroom. It consists of a 35 watt per channel Dynakit, A KLH FM tuner and 2 Maximus I speakers.)

The "perfect" room, designed for his new home three years ago, is 18 feet by 28 feet with a 9 foot high ceiling. The walls are wood, but with a difference—fiberglass is used in them as well as in the floor and ceiling for sound absorbing characteristics, though it's not too dead. He had the help of an acoustical engineer in designing it specifically for his speaker systems. "I'm gradually changing the acoustical environment," Close said, "adding shapes to the walls to break down the somewhat live conditions."

One of the most recent additions is the 100 pound Tarpon he caught on a 15 pound test line and spinning rig in the Bay of Florida last year. This and other game fish he's caught have been added to give some irregularity to the wall surface, as well as for wall decorations.

Though the room where the hi-fi equipment is used is the "family room," his wife and family of five children depend upon him for their hi-fi stereo concerts, since Dr. Close is the only one who operates the system.

Close's disappointment with much of the commercial tape recordings he can buy is what has led him to do his own recording. "It's the shortcomings of program material which are the limiting factor of extended range equipment," he says. "The tape duplicating companies haven't begun to use the best tapes available," he feels. So he makes his own transcriptions on the best tape available.

"I have many ¼ track commercial tapes, but I feel the quality could be definitely improved upon," Dr. Close commented.

He's recorded the carillon of the University of Chicago's Rockefeller Chapel, as well as its majestic organ; and back in 1962 he hired a hall and rented a concert grand Steinway piano so he could record piano music by Ralph Voltapek, the first winner of the Van Cliburn award. Those tapes are being aired over many of the nation's FM stations this year, along with a 15-minute in-

(Please turn the page.)

Four 70 watt mono power amplifiers are mounted openly on a shelf to assure good air circulation. Note the audio wall jack outlets under the shelf—mark of an audio hobbyist.

terview with the artist. And WFMT, the fine FM music station in Chicago, presented them in four one-hour recitals in December.

For his recording sessions, Close uses condenser microphones: Sony C-37A and Neumann U67. His friend Norbert Heckler, also an audiofan, often joins him on his musical safaris, helping to carry the heavy equipment and place the microphones.

Before one of these expeditions Close is assured that his equipment is in top performance by checking it with his laboratory of instruments, which he constructed himself. These consist of an audio analyzer, harmonic distortion meter, sine-square wave generator, oscilloscope, electronic switch and other allied equipment. These, too, are housed at a desk in the hi-fi room.

Through the years, since he began his serious interest in hi-fi in 1955 after his medical schooling was finished, Close has bought and sold many pieces of equipment to arrive at this ultimate system. Today he's an audiofan's audiofan.

Close thinks of himself as a semi-professional, since some of his sounds have been used on commercial recordings and for advertising commercials. His thunderstorm, for example, is on Dick Shory's *Happy Hits*. He has also recorded the Northwestern University Band for a nationally distributed record, as well as made numerous private records of various record groups.

He's now experimenting with 4-channel stereo with a friend who is an audio engineer. But he feels that the next challenges for him are in recording professional groups in really good acoustical environments. "That's hard to do because of union rules," he pointed out. But on occasion he's had that opportunity, by recording the ensembles so that they could hear themselves, and then turning over the tapes to them.

Dr. and Mrs. Close declined to put a price tag on their hi-fi system. "I haven't added equipment through ostentation," he emphasized, "but I think of everything I have as a tool, as the only way to evaluate tape recordings." He added that he doubted if 50 people knew of the extent of his equipment, yet, as a busy general practitioner, he knows hundreds of people in Chicago's North Shore suburbs. And while the hi-fi sytem is an avocation, it is to Dr. Close serious business.

From his other hobbies and interests, it's easy to see that he enjoys them all with the same intensity. He hunts and fishes, plays chess with his sons, and enjoys listening to short wave radio broadcasts.

The interest in radio and hi-fi actually came through his World War II assignment as a radio operator mechanic in the U.S.A.A.F. But it wasn't until after he'd completed his pre-medical training at Northwestern University, his medical study at the University of Illinois Medical School, and his internship at Cook County Hospital that he began to work with high fidelity.

Because of this background, the electronics of his system are as interesting and important to him as is the actual sound reproduction. "I enjoy working with my hands, too," he says "and building electronic equipment helps me to get away from the trials of medicine." He likes to build things from scratch and has built a number of kits for his friends.

What music does he play through the ultimate system when he listens for the joy of good music? It's the classics and semi-classics, and good jazz. "The only thing I don't listen to is rock and roll!" he exclaimed.

His tape library is so extensive that if he listened for 24 hours a day it would take him more than a month to hear them all, he says. But he's going to leave the cataloging of the collection to his children. In the meantime, he has the kind of memory which knows where every favorite tape is stored in the built-in shelves provided in the music room.

But even with that quantity of tapes, he's still looking for better and better sound reproduction to use in his system. He thinks it may some day come through a company which makes one to one tape duplications, and for the enthusiast such as he is, he hopes some day to interest a recording company in such a project on a custom order basis.

From the listener's point of view, what qualities are the outstanding of his ultimate system? Dr. Close believes there's an absolute lack of sound coloration and directionality. The acoustical quality of the right room and the full range electrostatic speakers, he observes, are significantly important to the good reproduction.

audioQuiz

Puzzles are fun, and they can be instructive. The quiz is a regular feature of AUDIOFAN. Readers are urged to send in quiz questions (and don't forget the answers!) AUDIOFAN will pay $5 for each one used. *Answers on page 17.*

1

In describing distortion, you sometimes come across phrases such as: first order distortion, second order distortion, etc. What do the different orders of distortion mean and what do they signify?

A continuing series of maintenance and service tips on hi-fi system components.

SIMPLE CHECK-UPS THAT KEEP YOUR SYSTEM TIP-TOP

HOLD ON TO HIGH FIDELITY

keep an eye on that recording level!

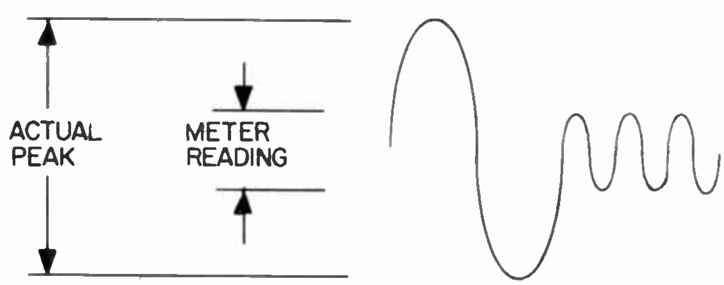
It's fine to squeeze out that extra margin of signal-to-noise ratio by setting the record level control on the high side. But you better watch your record level indicator closely throughout the taping session.

If you go past the maximum permissable record level you'll defeat your quest for better fidelity because distortion increases rapidly after that point (which is usually about 3% harmonic distortion). But even if you keep a sharp eye on the record level indicator you may find yourself plagued by noticeable distortion here and there on your tape. If this happens, and you're sure that you've sat on the level control and didn't allow any signal to bounce over the maximum recording level for more than a very brief moment, the problem may still lie with you, not the machine.

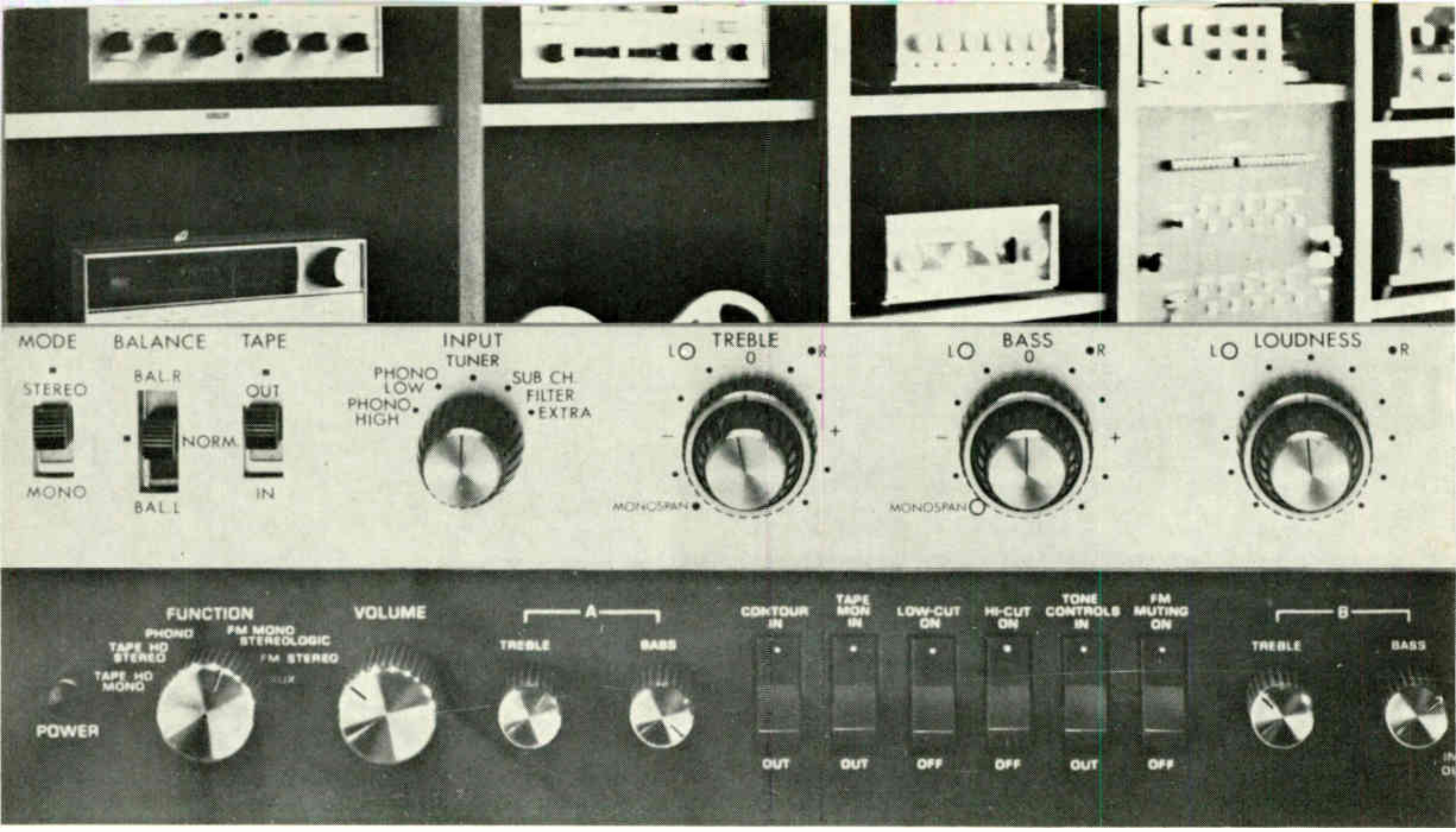
You may not fully understand how your record level indicator operates. This is particularly true with meter indicators, whether VU meters or VU types. The reason for this is that meters are mechanical devices and consequently exhibit some mechanical lag when activated by an electrical signal (which travels at the speed of light). As a result, a very strong transient signal may not be indicated on a meter.

To compensate for this, tape recorder manufacturers often set record level meters a bit ahead, so that a maximum recording level point, say, zero db, is reached by an indicator when the recording level is really a few db below it. But we don't know how much ahead meters have been set and, besides, some recorder makers don't follow this practice. So the only way to know where the true distortion point is reached, that is, where it really jumps once you pass the maximum point, is to experiment.

Based upon your experience, you can make good judgments concerning how high you can maintain your record level without running into objectionable distortion.



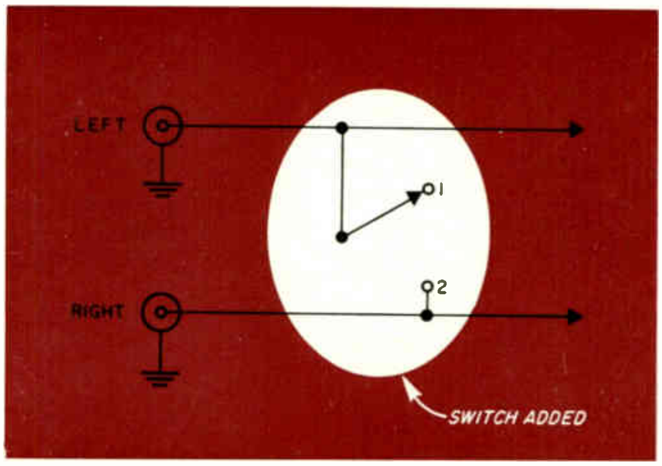
Record level meters may lie! A strong, momentary pulse, for example, may not register on some level meters. So to avoid recording with high distortion levels, become acquainted with how your particular meter reacts to transient signals.



A Hi-Fier's cockpit— the control section

power on; speaker systems selected; tuner function set; volume adjusted; treble and bass adjusted; sub-channel filter switched in; balance controls adjusted— all systems go!

In the event you have an older hi-fi unit that doesn't have an A channel plus B channel mode, you can get this facility by adding a wafer switch as shown here.



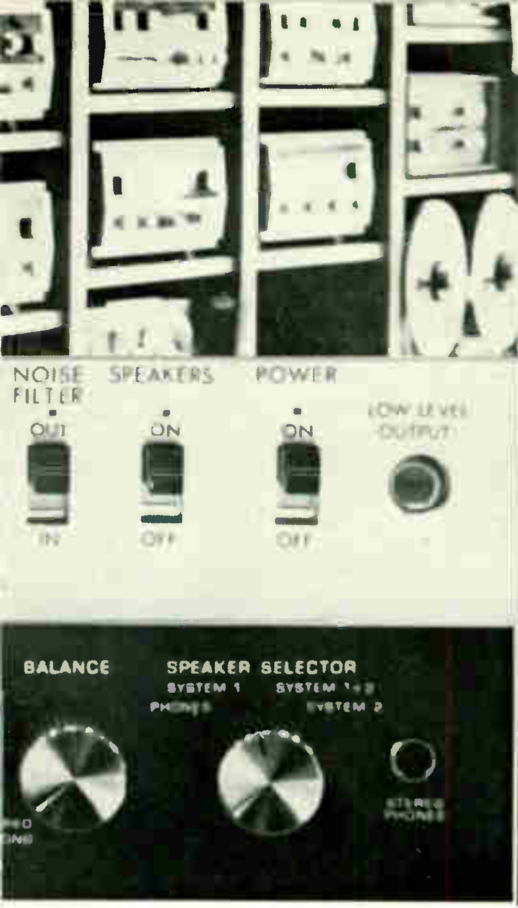
To those who are strictly "Outside Audio" the control panel of a top grade preamplifier may seem a bit overwhelming. But a good preamp, *we* know, needs a lot of features to allow a hi-fier to get the best sound in a variety of circumstances. On top of this, there are "extras" that make hi-fi that much more enjoyable.

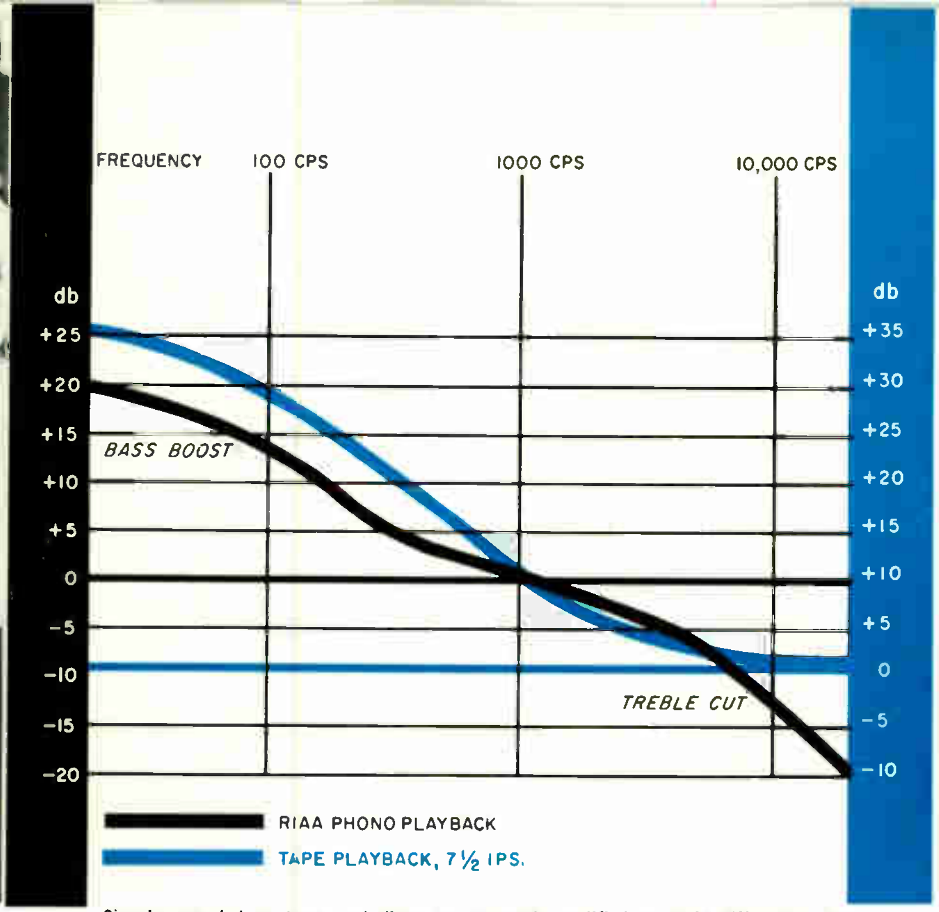
In between the inputs and outputs of preamplifiers are the controls and switches that permit hi-fiers to maneuver sound sources and frequencies as they wish.

The functions of the selector switch are pretty much self-evident, for example. Here is where you choose the source plugged into the preamp's inputs. Consequently, the selector switch is tied in with the input jacks which are usually located at the rear.

Another switch, the mode switch, allows you to switch from stereo to mono operation. It often has some additional modes that are seldom needed unless you experiment a great deal with different (mono) program sources: Channel "A" to both speakers, and Channel "B" to both speakers, for instance. With these modes, you could connect different mono sources to each side of your stereo input pairs, and get twice as many different programs selectable by switch as you normally do. Might be fun for a hi fi party where every guest brings his own tape machine—but you could start a trend this way, and the possibilities are plain frightening.

One thing it is nice to have in the mode switch is a paralleling of the two channels, for use when you play mono disks. You can, of course, play a





Signals recorded on tapes and discs are purposely modified to avoid difficulties that would impair fidelity. To get true reproduction, however, playback equipment must counter these changes, which it does through equalization circuits that modify the frequency spectrum as shown above.

mono disk in stereo mode, and everything sounds perfectly normal, with the same sound coming from both speakers. But if the two channels, and thus the two sides of the pickup, are connected in parallel for mono, signals from vertical motions of the stylus are cancelled out. You get rid of vertical rumble this way, and also of some distortion from "pinch effect," the tendency of the groove to get narrow, and push the stylus upward, in wide modulations. Some mono records sound considerably quieter, or considerably cleaner, or both, with a paralleled connection.

Some preamps parallel the two channels in the "mono" position, others give you an "A plus B" mode that does it. Look for one of these if you expect to play many mono recordings.

If you haven't got an "A plus B" position that puts the two sides of your pickup in parallel for mono playback (ask your dealer if it is not clear how this works on your unit) you may want to add a switch that does this right in the pickup input lead. Use a single-pole two-position wafer switch, not a toggle or all-metal rotary, to minimize hum pickup by the switch. Mount the switch *in the chassis* as close as you can to the pickup input connectors, again to minimize hum pickup. Connect it as shown in the drawing. One hazard you will have to keep in mind is leaving the switch in the paralleled connection when you play stereo records. With the switch in position 2, of course, the two channels are totally mixed together (zero separation) and exactly the same signal comes from both speakers. Stereo is completely gone, in this case!

Now, on to the tone controls. The first question is: should you have controls that control both channels at once, or separate ones for each channel? In every-day use, you want to adjust the two channels simultaneously, so a control that does both together is convenient. There *are* times for separate adjustment, for instance, if you are trying two different loudspeakers and need to make them match frequency-wise. In addition, separate controls allow you to compensate should one channel display different characteristics than another after a time.

But the truth is that such times come very seldom in the life of most hi-fi systems, so a single control for both channels is adequate for many of us. Probably the best all-around solution is the clutch-type control, in which the two channels can be adjusted together or separately.

tone controls

The electrical action of tone controls divides into two main types, the "hinge" and the "shelf." There is no overwhelming advantage of one over the other. Some technicians like the hinge type for professional use because it affects the gain in the middle of the spectrum less than the shelf type, but this is really of no consequence in home systems. The important thing is that you get *enough* change in the frequency characteristic so you can make your records and tapes sound the way you want them to. As a rough rule, a maximum bass and treble boost of at least 10 to 14 db and a maximum cut on both of at least 8 to 12 db would be fine.

(Please turn the page.)

(Continued from previous page)

In using tone controls, remember that recordings, both tapes and discs, vary tremendously from one to another in effective frequency balance, and the tone controls are there to make each one sound as close as possible to the way you would like it. So don't be afraid to use the controls freely to please your ear.

There are a considerable variety of *named* tone control circuits, one of the most popular being the Baxendall, called after its English inventor. The Baxendall does an excellent job, but is fairly elaborate and expensive. Another group comes under the heading of "feedback tone controls," which vary the response over the frequency band by varying the amount of feedback over the frequency range. To boost bass, for instance, the amount of feedback is *reduced* at the low bass frequencies, and as a consequence the amplification is increased at those frequencies. There are so many tone control circuit variations that a book could be written just on this subject. In some cases, the designer claims great advantages from putting the tone controls before, or after, some other stage, or from some other special configuration, but the crucial matters are more basic. They depend on designers' essential skills in amplifier design, covering such questions as, do the tone control circuits load a preceding amplifier too heavily, introducing distortion? Do they boost the highs, or lows, without affecting the middles too much? etc.

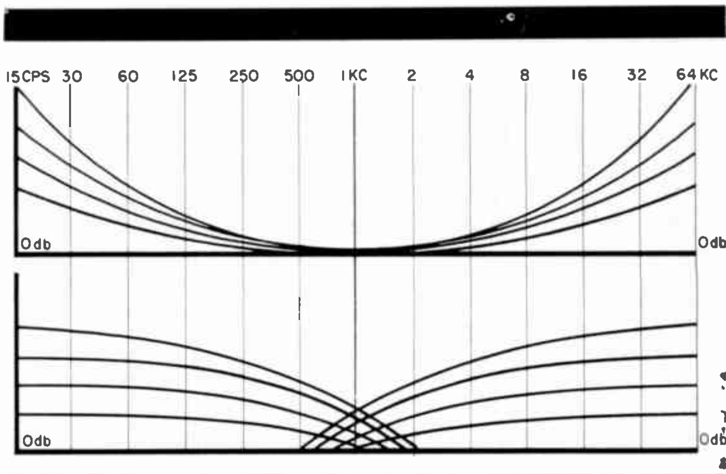
A last note on tone controls. The professional likes "step" controls, those using multi-position switches, because he can know just how much boost or cut, in decibels, he is getting in any switch position. But for home use, where tone controls are set by ear, the continuous control is usually satisfactory.

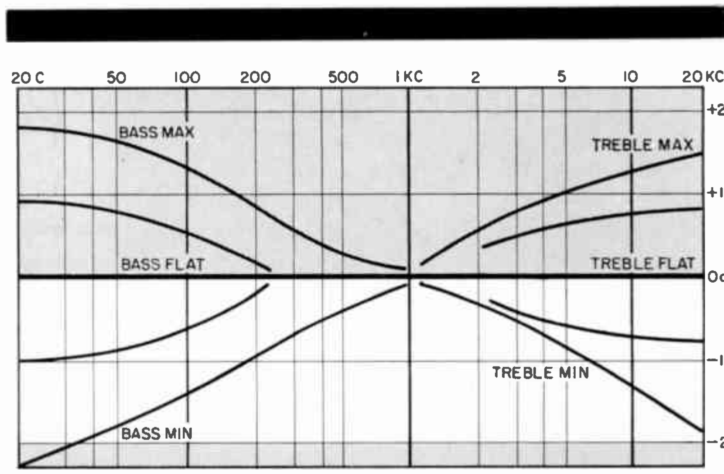
filter switches

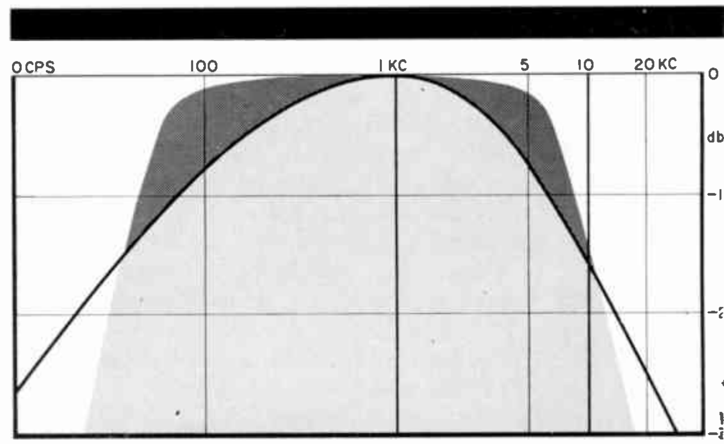
Related to the tone controls are the high and low pass filters, often called "rumble" and "scratch" filters. The rumble filter is supposed to cut bass sharply below some frequency, such as 60 cps, to cut out low frequency noise with a minimum of loss of bass frequencies in the music. This can be very handy with a noisy turntable, or a rumbly recording. Similarly, the "scratch" filter is supposed to take out a maximum of record surface noise, or tape hiss, and a minimum of the treble in the music.

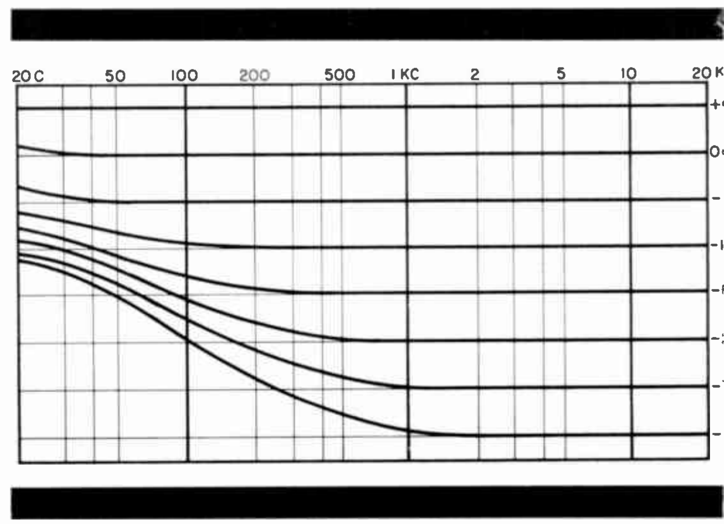
But cheaply-designed filters end up by taking out a lot of bass, along with the rumble, and a lot of treble, along with the hiss. A filter that really cuts sharply is fairly elaborate, and can't be done inexpensively.

It takes a filter circuit using quite a few resistors and capacitors to do even a passable job, and for *really* sharp cut-off you need a circuit that includes a coil and capacitor circuit, as well as a number of resistors and other capacitors. In other words, a circuit that makes use of the sharp frequency discrimination of capacity and induc-









There are innumerable types of bass and treble tone control designs. One type gives the tone control action shown at left, where levels at 1 kHz are not raised when highs or lows are boosted. Below it is another type that gives 'shelf' action. That is, levels are raised somewhat during boosting.

Simplified curves of tone control characteristics are shown here. Many hi-fi manufacturers include such performance charts in instruction manuals accompanying their equipment. As an example of how this chart is used, setting a bass control to maximum will give you about a 12 db boost at 100 Hz, a 16 db boost at 50 Hz, etc.

Hi-fi filter switches are supposed to cut-off audio in certain frequency spans to eliminate undesirable sound such as rumble, scratch or hiss. This cut-off occurs gradually, sometimes too gradually, taking along some music with it. Examples of good and bad high and low filter characteristics are shown. Note that the slow, sloping curve cuts out much sound (dark gray area) at 100 Hz, though the goal is 60 Hz rumble.

As sound level is decreased, we lose some hearing sensitivity to bass frequencies. To compensate for this, loudness controls are often employed to automatically boost the level of bass frequencies as gain is lowered. If you don't have a loudness control on your control amplifier, you can prevent some bass loss by turning up your bass tone control.

tance used together. Such circuits are quite expensive, compared to the ordinary "tone control" type which simply cuts either highs or lows over a very wide range.

A sharp filter, however, has the ultimate disadvantage that it might introduce some phase and transient distortion. In the case of a very rumbly or hissy recording, the distortion caused by the filter is many times less obtrusive than the noise, so use of a sharp filter may be an excellent choice for maximum fidelity. But if you are not bothered by the low or high frequency noise that the filter is supposed to reduce, don't use the filter: this is the rule to follow.

There are some frills on front panels that make life easier, but are not *absolutely* needed. One is a speaker phase reverse switch. On the surface, this seems to be handy indeed. When you install your system you will have to get speakers in phase, and this has usually been done by reversing the connections to one speaker—assuming the original connection put them out of phase. So why not a switch on the panel that saves disconnecting and reconnecting a speaker?

Since the job is generally needed only once, on installation, a panel switch is not really a necessity and may even be a disadvantage because it can inadvertently be thrown the other way. But if you are trying new speakers frequently a switch could save trouble. So each hi fi user will have to decide how important this is to him. But if a preamp has everything else you want, then don't hold up because it lacks a phase reverse switch: it's never *that* important.

Similar reasoning applies to the "loudness control," a very old idea in hi-fi amplifiers. Briefly, the loudness control boosts bass automatically when you turn to low volumes, to offset the ear's lowered sensitivity to bass at low volumes. That sounds fine but an *exact* compensation for the ear's characteristic would require a very elaborate control, with different amounts of boost for different loudness levels on the record and different acoustic conditions in the recording. An approach to this has been made in some amplifiers with a separate "loudness contour control," plus a volume control.

Since the whole purpose of loudness compensation is to preserve a pleasing amount of bass at low volumes, why not simply set the volume where you want, and then set the bass boost to a pleasing level? It can be argued that without loudness compensation you would run out of bass boost on some recordings that are very weak in bass. And adjustment isn't automatic.

A variety of other switches are often used today, including: tape monitor, tape function, equalization, speaker selector, and power switches, among them. A low level stereo output jack is generally on front panels of most modern units, too.

In sum, today's control panels give audiofans more operating conveniences and greater flexibility than ever before.

audioQuiz

ANSWERS

The order of distortion refers to the manner in which frequencies other than the original frequency fed to the input of an amplifier are produced at the output of an amplifier. For example, if a 1000 Hz signal is introduced to an amplifier, the output frequency should ideally be only an amplified 1000 Hz signal. We know, however, that some energy in the form of other frequencies appear, though in small proportion to the original frequency.

Energy as a second harmonic of 1000 Hz might appear, for instance. The frequency would be 2000 Hz as a result of adding 1000 Hz and 1000 Hz. This new (and undesirable) frequency is the first order distortion. Adding another 1000 Hz to the original frequency produces 3000 Hz, which is the third harmonic of 1000 Hz. Energy produced at this frequency is the second order of distortion, etc.

Other possibilities are presented when more than one tone is involved. The sum and difference of two tones, say 1000 Hz and 500 Hz, could produce both harmonic distortion and intermodulation distortion. First order intermodulation distortion would be energy produced at 1500 Hz (sum) and 500 Hz (difference). Second harmonics of these frequencies may, in turn, produce second order intermodulation distortion, ad infinitum.

As the order of distortion progresses, energy fortunately diminishes, so the biggest concern is with first order distortion. Other orders are additive, of course. All the orders of harmonic distortion are generally combined as total harmonic distortion in specification, which might be a THD of 0.1%, as an example.

how to attach phono plugs

Into every hi-fier's life comes the day when he switches on his system and is greeted by an ear-bursting hum, silence, or a crackling sound, any of which can be caused by a broken or poorly connected phono plug. This is the plug that transfers audio signals from one component to another (say, a phono cartridge to a control amplifier's input jack).

The plug-to-shielded wire connections break from time to time for some inexplicable reason (which Sherlock Holmes would undoubtedly trace to a maid's dusting efforts, a hi-fier eagerly removing a plug by pulling on the shielded wire instead of grasping the plug itself, or some such cause). Should it happen to you, you can restore your stereo music quickly with just a little effort backed up by some know-how. Simply follow the step-by-step photos shown here.

Starting from scratch with a new phono plug (you should always have a few in the house; they're certainly inexpensive enough), hold the new phono plug against the shielded wire as shown in Fig. 1. Allow about ½" space from the tip of the plug to the end of the shielded wire, which should be snipped off cleanly.

About ¼" *below* the length of the plug carefully nibble around the shielded wire's outer insulation, usually plastic, with a diagonal cutting plier. See Fig. 2. When this has been done, slide off the outer insulation, as illustrated in Fig. 3. You might have to use a little pressure with the cutting pliers.

With the insulation removed, the cable's metal braiding is revealed. The braiding, which is the grounding lead, surrounds the cable's inner insulation which, in turn, covers an inner conductor. This central wire is the second conductor of the shielded wire. So though a shielded cable's construction differs from a conventional lamp wire's construction, they both have two leads or conductors.

The next step in preparing a shielded cable to accept a phono plug is to convert the metal braiding into a wire lead. This may be done in two ways. One method is to part a small section of the braid with a scribe or ice pick, burrow into the opening to below the inner insulation without damaging the insulation, and pry the inner insulation and its central wire up through the hole.

The foregoing might be the "professional" way, but unless you're a handy sort it is fraught with hazards such as damaging the inner insulation, which could result in a short circuit between the outer conductor (the metal braid) and the inner conductor (the central wire), or hurting yourself by slipping while pulling up the insulation with the scribe or pick. A safer, easier way is shown in Figs. 4 and 5.

Here the metal braid is separated by inserting the pick into the braid near the end of the wire and working the tool to unbraid the ground wire. This operation is repeated until all the exposed ground wire is unbraided. When completed it'll look like the photo in Fig. 6. Then twist together the outer grounding the individual wires together to form a strap, as seen in Fig. 7. This is the completed grounding lead. Now for the other lead.

About ⅜" up from the outer insulation gently cut into the inner insulation. Use a careful squeezing technique with the cutting pliers, moving the cutting edges in a sort of circular motion. Then strip away the inner insulation, leaving the central wire exposed, as shown in Fig. 8.

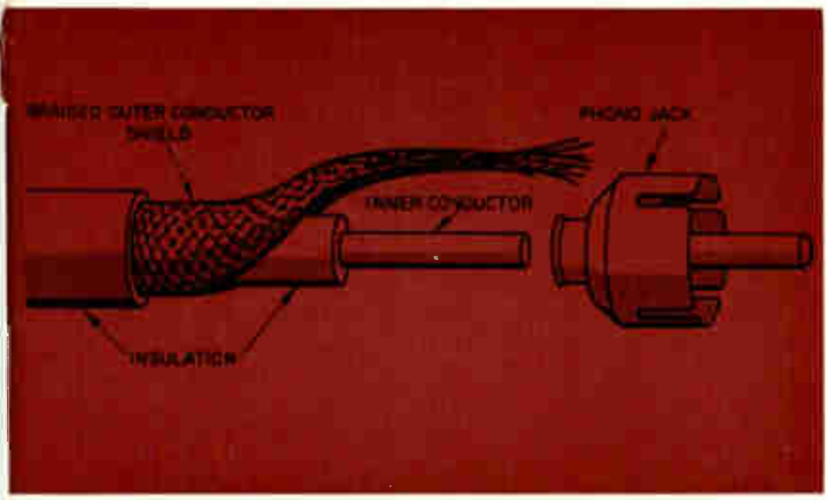
Now you have to prepare the two wires for attachment to the phono plug. A vise comes in mighty handy now. The remaining steps can be accomplished without one, but it becomes a bit frustrating in spots. In addition, everyone seems to mildly burn their fingers at one point or another when soldering without benefit of a vise

Using a vise, gently lock the cable into its jaws. Now "tin" the inner conductor and the outer conductor. Tinning simply means preparing the conductors for soldering onto another part by applying a thin coating of solder (which should always be rosin-core solder), as shown in Figs. 9 and 10. Now slide the tinned inner conductor into the phone plug, as illustrated in Fig. 11. Part of the bare central wire's length will extend past the phono plug's hollow tip.

Secure the phono plug's tongue, where the inner wire extends, in the vise. Be sure to bend the protruding wire to prevent the whole cable from sliding off the plug. Now proceed to tin one side of the shell's neck, as shown in Fig. 12. This will make it easy to secure a good solder joint. Now, holding the twisted grounding wire with long-nose pliers so that the wire presses against the tinned section of the phono plug, solder

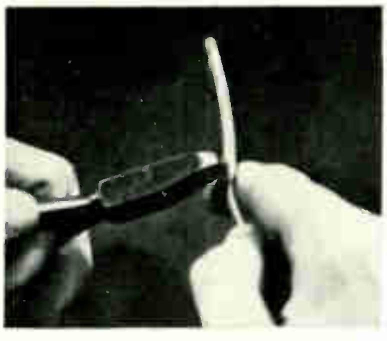
CONSTRUCTION PROJECTS

The drawing illustrates phono connector cable parts ready to be soldered. Photos below give step-by-step instructions.



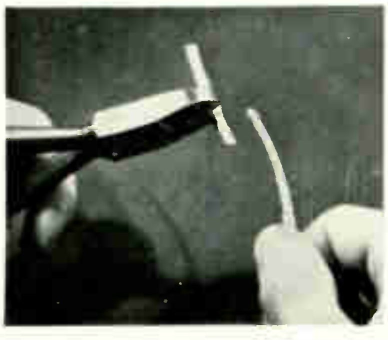
1

Hold the phono plug against the cable as shown, leaving some space between the tip and the end of the cable.



2

About a ¼" below the bottom of the plug, nibble around the shield cable's outer insulation.



3

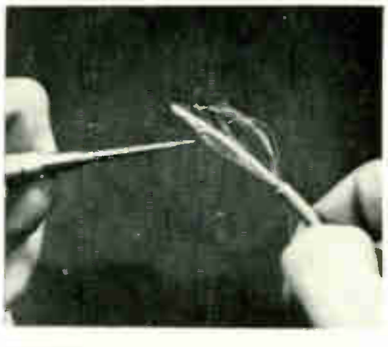
Slide off the outer insulation when this has been done. Be careful that you don't cut through the insulation and damage the shielded wiring below it.

4

An easy way to convert the shielded wire into a wire lead (the ground lead) is to use a pick or scribe to separate wires that make up the shielding.

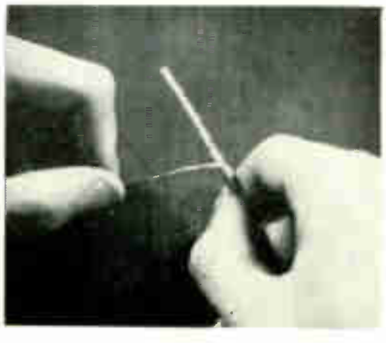
5

Work the scribe down from the tip of the shielding until all the exposed shielding wire is unbraided.



6

The unbraided shielding looks like this.



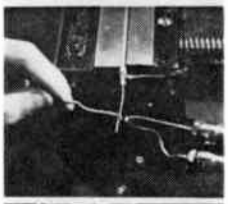
7

Twist the unbraided shield wires together and you've got a single lead for the ground wire.

Photos reprinted courtesy of Bolex Reporter Magazine

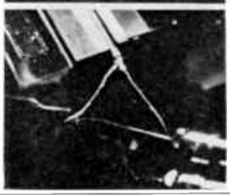
8

Strip off the insulation covering the center wire.



9

Placing the cable in a vise, tin the inner conductor.



10

Now tin the outer conductor (twisted shield wire).

11

Push the center conductor through the opening in the phono plug tip.

12

Place the plug in a vise and tin one side of the shell.

13

Solder the unbraided shield wire to the shell. Snip off excess wire.

14

Reverse position of plug in vise and heat some solder so it flows into tip opening.

15

After cooling, cut excess wire.

16

Voila! The completed cable.

(Continued from previous page.) the strap to the shell. Hold the grounding strap firm for 10 seconds or so after removing the solder iron or solder gun to give it a chance to cool off. If you allow the strap to shift, you might suffer a "cold" solder joint which could give you intermittent connection problems later on. Snip off the excess ground strap with the diagonal cutting pliers. See Fig. 13.

Reversing the phono plug's position in the vise, with the tongue end secured as shown in Fig. 14, heat the tip of the plug while feeding a small amount of solder into the heated opening (where the central wire protrudes). The melted solder will flow inside the opening and solder the inner conductor to the inside of the tongue. After a cooling period of 10 or 20 seconds, snip away the excess inner conductor, as in Fig. 15.

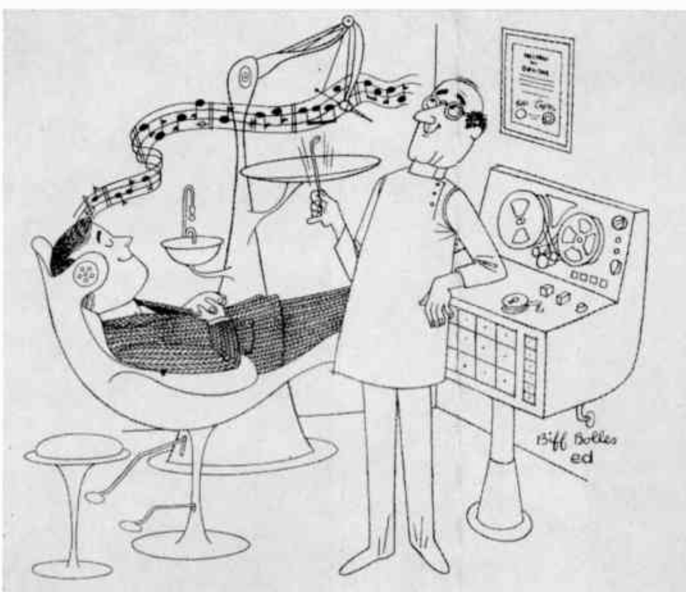
Fig. 16 shows the finished phono plug and shielded cable, all ready for its high fidelity task.

It must be admitted that it's easier to go out and buy a new phono plug cable assembly than attach a plug to a cable, but if you don't want to forego your music late at night, on a Sunday, or a holiday (and isn't that when breakdowns always occur), you have no choice. Besides, you'll save money this way.

MUSICAL DENTAL CHAIRS

"Gee, I think I'll go to my dentist for a checkup. I hear he's got the latest Maria Callas records."

This might not be as far-fetched as it seems. Music plays a large part today in "treating" dental patients. For example, dentist-office jitters are often allayed by pleasant music piped into the room from some unknown, but ostensibly friendly, source.



Many dentists have adopted the idea of "musical chairs" to minimize discomforts of dental drilling and probing. For example, some dentists find that hi-fi music fed into stereo headphones worn by their patients tend to relax them while they're working on their teeth. More important, the music distracts a patient's attention from the discomforts of dental drilling and probing. Another sound device used with some success depends upon "white noise" or "waterfall" sounds from a set of headphones to divert a patient's attention from dental work. Sound level is often controlled by the patient himself . . . the louder the sound, the more distracting it is said to be.


New from Garrard
Model SP 20
at only $37.50
an excellent 4-speed manual record playing unit with high fidelity features. This compact, efficient player is recommended for basic music systems and quality audio-visual applications.
Semi-counter-balanced arm with adjustable stylus pressure
Full size weighted turntable
Trip of DuPont Delrin®... track as light as 2 grams
Size—Compact 14⅜" left to right, 12½" front to rear, 3½" above and 2⅛" below motor board. Fits Garrard B10 series bases.
Interchangeable plug-in head, for any cartridge
Automatic return of arm to rest, and shut-off after play
Motor designed and built entirely by Garrard
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WORLD'S FINEST

WHAT'S GOING ON

Microphones

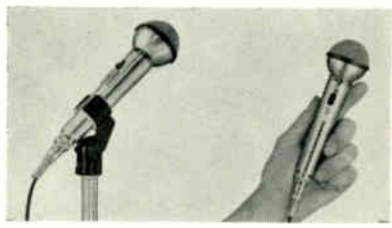
With few exceptions, microphones provided with tape recorders can't bring out the best recording capabilities of the machine. And there's no reason why they should since they generally cost a pittance compared to ones that can produce wide, smooth frequency responses. Consequently, hi-fiers who are at all serious about live recording buy mikes separately. Here are some moderately priced ones you might consider.

ELECTRO-VOICE E-V's Model 635A broadcast microphone is ideal for home recording, too, advises the manufacturer. Its below $50 price places it within easy reach of

many recording hobbyists. The unusually light (only 6 ounces), slim (3/4" diameter) microphone makes it practical for it to be used as a lavalier mike as well as a hand-held, desk or floor-stand mounted microphone. Among its interesting features is an internal shock absorber which is said to greatly reduce impact noises often incurred by people not especially skilled in microphone handling, and a 4-stage filter that eliminates pops and wind noise as well as guarding against dirt and moisture. The low impedance microphone's response is 60 to 15,000 Hz, while output is -55 db.

ERCONA The Swedish PLM EK-61A condenser microphone marketed by Ercona breaks a lot of conceptions concerning this type of microphone. It's unusually small (measures only 2-11/16" x 11/16"), featherweight light (weighs only 1-1/4 ounces), and considering that most condenser mikes cost upwards of $200, its cost is low ($99.50 + $19.50 for a matching battery-operated power supply.) According to specs, frequency response is a linear below 30 Hz to above 18,000 Hz. The ominidirectional EK-61A microphone comes with 10 feet each of signal and power supply cable and a clamp-on mike stand adapter. A cardioid version is available for $109.50, as are other AC and battery operated power supplies.

SHURE A new ominidirectional dynamic microphone, the "Spher-o-Dyne," features a uniform frequency response from 40 to 11,000 Hz. This makes it useful for a wide range of applications where pinpointing a source, say, a singer's vocalizing without recording audience noises, is not required. (A unidirectional or cardioid microphone would be desirable in this case). The mike, whose name was coined from its spherically-shaped front, has a built-in wind, breath, pop filter, that allows close-to-mouth operation without recording ear-bursting breath sounds. Among its other features are a built-in On-Off switch and a sep-



arate, adjustable swivel adapter which enables a user to tilt the mike 90° when mounted on a stand. The microphone can be quickly removed from the adapter for hand-held use, which its small size and light weight (11 ounces) invites. There are two models: Model 533SA ($50.00), a high impedance unit, and Model 533SR ($47.50), a low impedance unit (desirable when long cable lengths are required). Both models are supplied with detachable 15 foot cables.

Headphones

If there's one hi-fi item that people come back for shortly after buying a hi-fi system it's headphones! A growing number of audiofans, in fact, buy 'em along with the rest of a system because they recognize that headphone listening and loudspeaker system listening each provide distinctly different aural illusions, both satisfying in their own way. Here are a few stereo headphones that have come to our attention recently.

AMPEX Ampex expands its line of audio tape recorders with its first headset model, the new Model 140. The stero headphones incorporate sensitive 3-1/3" dynamic transducers in each ear piece. The molded sponge plastic ear and head cushions make for a lightweight device, which enables a user to wear the phones with comfort for extended periods of time, says the manufacturer. The phone set's headbands may be adjusted to any head size. Each 140 has a single two-way jack and

accessory cord to allow the user to connect the headset to either single or double stereo outputs. (many amplifiers have a front panel single jack that produces twin channel outputs). Priced at $36.95.

SHARPE A combination noise attenuation cup and cavity cup divider insures 20 to 20,000 Hz ± 3 db performance from the new Sharpe Model HA-660/Pro stereo headphones, says the manufacturer. The headphones come with individual volume controls, which makes it convenient to adjust listening volume at a distance from amplifier controls. The phones feature liquid-filled noise attenuating ear seals, which, if you've never tried them, gives a tight yet comfortable seal between phones and your ears. Together with a dual slide nylon insert headband with padded cushion, sound leakage is prevented. The drivers, which can accept up to 2 watts input power each, are individually fused to protect them from overloads. Some specs are: Noise attenuation is 40 db at 1000 Hz; 8 ohms impedance per phone (adaptable to 4, 8 and 16 ohm outputs). Priced at $60.

1966
coming events

FEBRUARY 18 to 20 (Friday to Sunday)
Philadelphia High Fidelity Show
BENJAMIN FRANKLIN HOTEL, PHILADELPHIA, PA.

MARCH 30 to APRIL 3 (Wednesday to Sunday)
Los Angeles High Fidelity Show
HOTEL AMBASSADOR, LOS ANGELES, CALIF.

APRIL 20 to APRIL 24 (Wednesday to Sunday)
San Francisco High Fidelity Show
PACIFIC AUDITORIUM, SAN FRANCISCO, CALIF.

APRIL 25 to APRIL 28 (Monday to Thursday)
Audio Engineering Society Convention
HOLLYWOOD ROOSEVELT HOTEL, LOS ANGELES, CALIF.

inquiring reporter

(IF YOU HAVE A QUESTION YOU WOULO LIKE OUR INQUIRING PHOTOGRAPHER TO ASK, WRITE US YOUR SUGGESTIONS.)

PLACE: NEW YORK HI-FI SHOW

QUESTION: WHAT COMPONENTS IMPRESSEO YOU MOST?

IRVING GOLOMB, ATTORNEY, 20 YEARS HI-FI INTEREST: I WAS GREATLY IMPRESSED BY SMALL SPEAKERS. I'VE ASKED FOR THE LITERATURE AND I MAY CHANGE OVER TO THEM. THEY HAVE A PERFECTLY BEAUTIFUL SOUND. YOU SEE, I PLAY THE VIOLIN. I FIND MOST SPEAKERS WHICH BOAST OF THEIR HI-FI CAPABILITY NEVER CONVINCE ME THAT THEY'RE DOING WHAT THEY SAY THEY CAN DO. NEVER SOUND QUITE RIGHT.

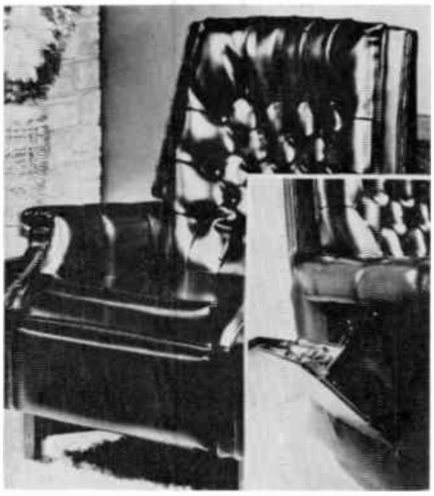
D. RANDOLPH JOHNSON, GRADUATE STUDENT, TEN YEARS HI-FI INTEREST: TAPE RECORDERS AND THOSE BIG SPEAKERS. THOSE THINGS HAVE ABSOLUTELY AMAZING SPECIFICATIONS. AND DON'T FORGET THOSE PATRICIANS WITH THEIR 30-INCH WOOFERS. I FOUND SOME NEW PRODUCTS BUT I DON'T SEE A CARTRIDGE THAT WILL MAKE ME CHANGE FROM MY PRESENT ONE. THEY'RE PROBABLY MAKING COMPARABLE CARTRIOGES, BUT I'M GOING TO STICK WITH WHAT I HAVE FOR NOW. WHAT INTERESTS ME MOST IS TO SEE WHAT EQUIPMENT A MANUFACTURER USES TO DEMONSTRATE HIS PRODUCTS: FOR EXAMPLE, WHAT MAKE OF TURNTABLE AND SPEAKERS IS BEING USED TO DEMONSTRATE A CARTRIDGE.

JOHN ROHALY, ACCOUNTANT, 12 YEARS HI-FI INTEREST: OF COURSE, WHAT IMPRESSED ME MOST IS THE 0.33 IPS TAPE RECORDER. IT MEANS THAT YOU CAN GET AN AWFUL LOT OF MUSIC ON ONE TAPE. AS A MATTER OF FACT, I'M HAPPY TO SEE THE GENERAL TREND TOWARDS TRANSISTORIZED TAPE RECORDERS. BEYOND THAT I FOUND NOTHING SPECIFICIALLY IMPRESSIVE. THERE IS A GENERAL IMPROVEMENT IN PRODUCTS WHICH I HAVE FOLLOWED OVER THE YEARS.

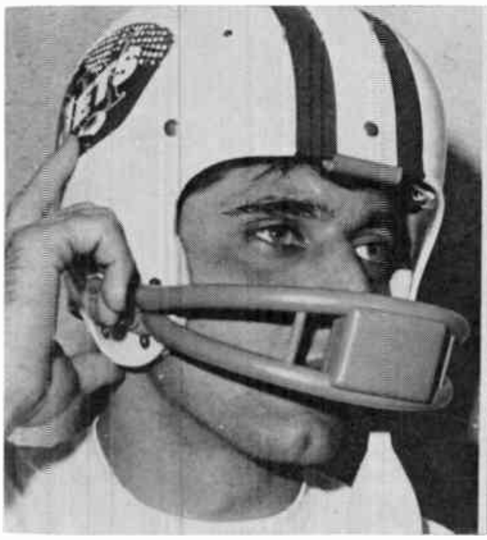
the Model T Ford and old phonographs had one thing in common—a crank.

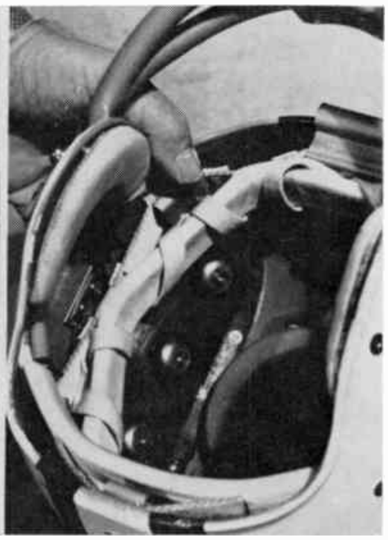
Here are some muscle-powered relics collected by Benjamin Electronic Sound Corporation. The company offers a new Benjamin stereo unit for every ancient model device it accepts. The collection will be used for window displays and by schools and colleges in an educational program.

here's a reclining lounge chair that's wired for music.



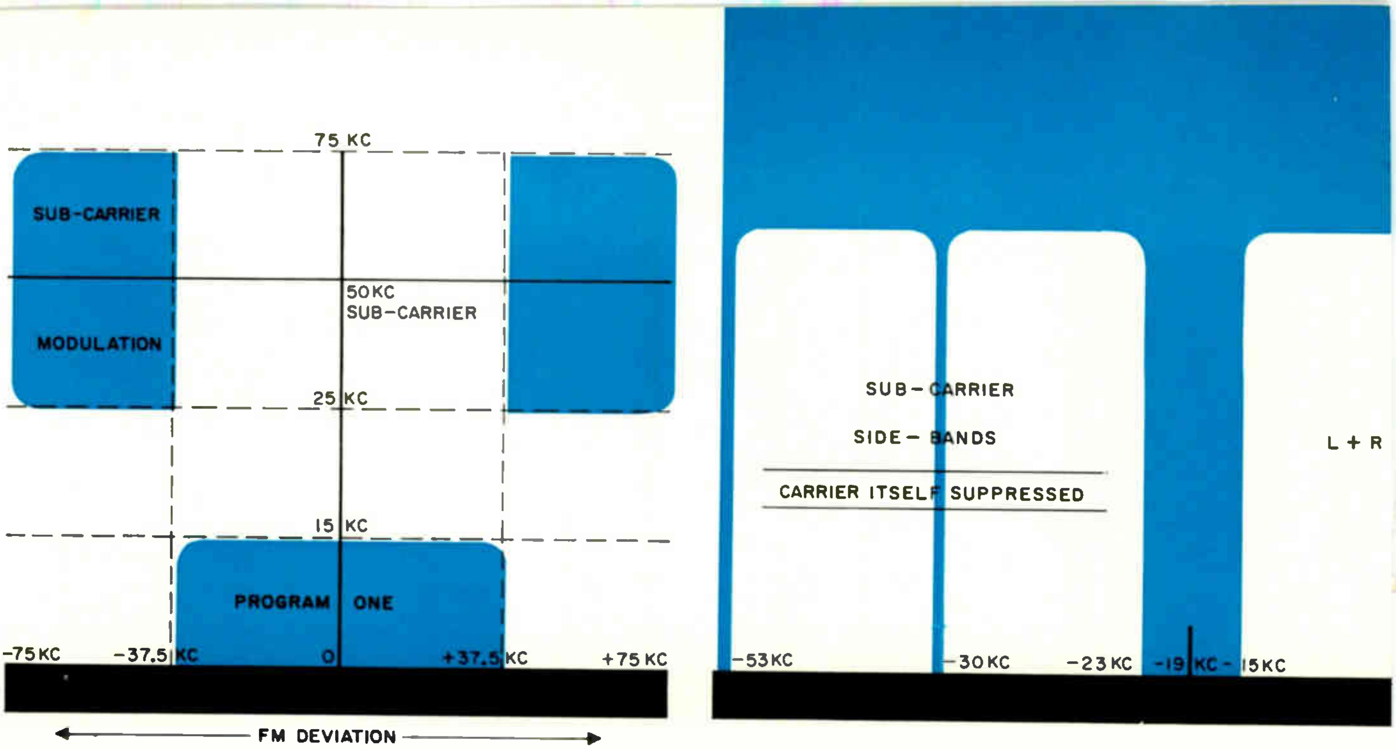
Berkline Corporation's "Stereo-Lounger" conceals a self-contained stereo tape player in a tilt-out compartment on one side of the chair (see inset). A power line cord to be plugged into a standard wall outlet is located at the rear of the chair. Speakers are hidden in the two wings of the chair (what, no headphones!). The music system is not what we audiofans would consider high fidelity by any means (rumor has it that it's a Lear Jet continuous loop tape cartridge player), but the idea of a fully concealed music system with fingertip operating convenience combined with a chair that reclines and swivels is appealing.





electronic lung power helps quarterbacks.

Joe Namath of the New York Jets models the football team's new "electronic lung power" helmet, which featnres six tiny speakers mounted inside. The helmet actually contains a miniature transistorized public addess system to boost the quarterback's voice to some three times londer than a TV set's audio going full blast, it's reported. The speaker system, designed by Jensen Manufacturing engineers, is designed to reproduce voice frequencies (900 to 3000 Hz). As shown in the photograph, three speakers are mounted over each of the quarterback's ears. Small vents allow the sound of the quarterback's amplified voice to be distributed outside the helmet area. To use the electronic system the signal caller speaks into a small microphone fastened firmly to the inside of his face guard. In the event you're wondering if the rest of the team is exposed to tortuous sound levels when they're in a huddle, they don't. The quarterback simply switches the system off.



THE COMPATIBLE FM SIGNAL

how to get two channels from a single FM broadcast frequency

by LEONARD FELDMAN

FM Stereo broadcasting has been with us, now, for over four years. Though many enthusiasts have a pretty good idea of how one can squeeze both the left and right signals of a stereo program into a single record groove and subsequently recover the information, few audiofans have been able to figure out the more complex problem of getting two radio programs (the left and right channels) out on a single FM station and receiving both channels of information with a single tuner, tuned to only one frequency.

Before we tackle FM stereo, however, let's review the differences between AM (lo-fi) and FM (hi-fi) broadcasting. As you may already know, in FM broadcasting, the amplitude or size of the radio waves do not vary. Information is transmitted by periodic changes in frequency, corresponding to the tone or program which is to be transmitted. This is illustrated in Fig. 3 (Page 28). At the top of the diagram is shown a pure sine-wave tone of, let us say, 400 Hz. The lower portion of the diagram shows (in exaggerated form) how the radio station carrier wave will move higher and lower in frequency at a rate corresponding to this tone.

There are two, main advantages to this type of transmission. First, since noise or static generally is represented as additions to or subtractions from the *amplitude* of the radio carrier, the frequency variations used to transmit the desired program material remain unaffected by these random sources of static. The receiver can be insensitive to amplitude variations (and hence, insensitive to static) and responsive only to variations in frequency. Secondly (and this advantage is not inherent in FM per se), because there is more "room" at higher frequencies, FM stations have been asigned sufficient bandwidth or frequency spectrum to permit full-fidelity transmission, with all audio information between 50 Hz and 15,000 Hz included.

In contrast to this, the AM band is so limited that most stations are restricted to a high frequency limitation of only 5000 Hz—hardly hi-fi!

early FM stereo

As early as the late 1940's, the late Major Edward Armstrong foresaw the need for stereo ("binaural," as it was then called) radio. His basic approach for transmitting two different programs over a single FM channel is shown in Fig. 1. Since there is a total of $\mp$75 kHz available for each channel to deviate (or, FM modulate) about its center frequency, he proposed that one program channel use up perhaps 50% of this deviation to transmit regular music or voice. The remaining available 50% was to be modulated by a sub-carrier, a super audible frequency in the range of about 42 kHz.

Given a normal receiver, the listener would not detect the presence of this super-audible fre-

Fig. 1
A two dimensional representation of main and sub-channel modulation of the FM carrier frequency as foreseen by Major Edwin Armstrong, the inventor of the FM (frequency modulation) system of broadcasting and receiving.

Fig. 2
The elements of the approved FM stereo composite signal are illustrated here. The left plus right (L + R) channels modulate the main carrier frequency directly. Left minus right (L — R), however, AM modulates the sub-carrier, 38 kHz, resulting in poorer signal-to-noise ratio for stereo. That's why an outdoor FM antenna is recommended.

quency. This sub-carrier could, however, be modulated in turn by a second program source. In other words, the modulated sub-carrier and the program 1 material would *both* modulate the main radio-frequency carrier. A listener equipped with an FM receiver plus a decoder could receive program 1 in the normal fashion and could, by means of the additional decoder, receive program two, as well. Obviously, programs 1 and 2 could be the left and right channels of a stereo program if desired.

This system is actually in use today, but not for stereo broadcasting. Instead, the system described is used for simultaneous transmission of background music heard in restaurants, public places, etc. Stations engaged in public programming are permitted, by this means, to rent out to subscribers commercial-free background music, rendering their economic position more secure.

As for stereo, however, the objections to this system are fairly obvious. For one thing, a listener equipped with only a conventional FM receiver would hear only *one side* of a stereo program. Then too, for somewhat technical reasons, the signal-to-noise qualities of program 1 would differ markedly from the signal-to-noise characteristics (or static free qualities) of program 2. Finally, if 50% of the deviation capability per station were devoted to program 2 (by means of the inaudible sub-carrier), the monophonic listener would suffer a 50% decrease in loudness and signal-to-noise when listening to the program monophonically.

compatible stereo FM

All of the above objections were overcome in 1953, when a disciple of Armstrong, Murray G. Crosby, patented a system of compatible stereo broadcasting—one in which the monophonic listener isn't even aware that stereo is being broadcast unless an announcer tells him so.

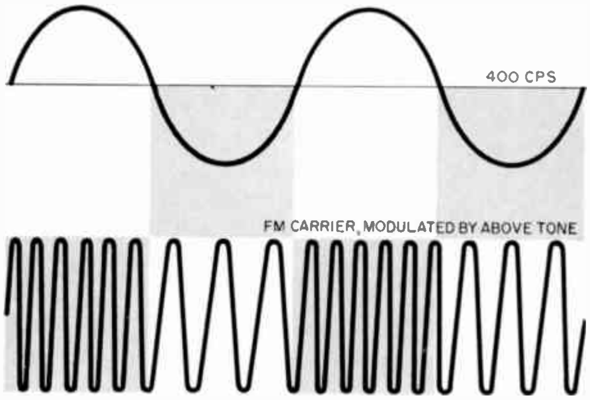
To begin with, the material of program 1 (left channel) and program 2 (right channel) are electrically added together to form left (L) + right (R). But, L+R is immediately recognizable as the monophonic equivalent of the entire program, since it is the sum of both the left and right channel information. This L+R signal, said Crosby, should be used to modulate the radio-carrier at a level of 50% of the total permissible deviation. Next, form a signal by *subtracting* the R program from the L program electrically (L—R) and use this signal to FM modulate a super-audible 50 k Hz carrier, which in turn will modulate the main carrier for the remaining 50% of capacity.

At the receiving end, the normal FM mono receiver would uncover the L+R signal (mono) directly, which would satisfy the mono listener. For the stereo listener, a decoder circuit would recover the L—R information and perform the following two alge-

(Please turn the page.)

Fig. 3

In FM, the frequency of a carrier signal is altered by program material, in this case a 400 Hz signal. Note that the height of the carrier signal is constant. With AM radio, the frequency is constant and the height is varied by the modulation signal.



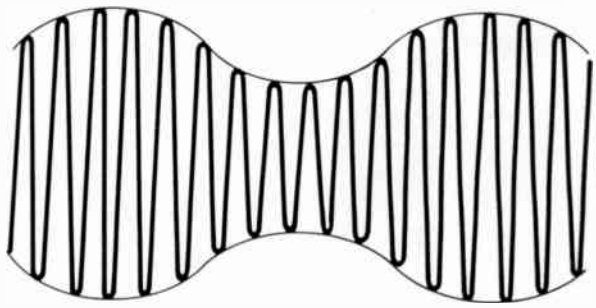
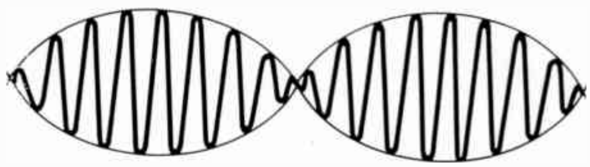


Fig. 4

Here is how the L-R audio in an FM multiplex stereo system is converted to suppressed 38 kHz carrier sideband information. The top signal illustrated is an AM-modulated sub-carrier. After removing the sub-carrier, the remaining sideband products, shown below, further modulates the main carrier.



(Continued from previous page)

braic maneuvers with it:

(L+R) + (L–R) = 2L
(L+R) – (L–R) = 2R

Thus, separate and independent L and R programs would be recovered for application to the convention stereo amplifier and speaker pair.

the FCC decides

In April 1961, the Federal Communications Commission (FCC), after much deliberation, approved a system for stereo broadcasting which has many of the elements proposed by Crosby, with variations proposed by General Electric and Zenith Radio Companies.

One of the objections raised by the FCC in connection with the "pure" Crosby system was the fact that the mono listener would be getting an L+R (mono) signal only 50% as loud as the unmodified signal, because of the 50% deviation proposed by Crosby. A diagram of the final makeup of the approved stereo signal will help explain the differences between it and the Crosby system. It is shown in Fig. 2. While L+R is still used to modulate the main carrier directly, L–R is used to AM modulate the sub-carrier (whose frequency is 38k Hz). This modulated sub-carrier is then stripped of its carrier, so that only the modulation products (known as sub-carrier sidebands) remain.

This process is shown in Fig. 4. It is this side-band information which further modulates the main carrier, so that L–R information travels along with L+R information, much as in the Crosby proposal. Because of the inter-relationships between the L+R and L–R sideband information, it so happens that each of these components can be used to modulate the main carrier to 90%. It's a case where 90% + 90% still equals only 90%!

The secret is that what is actually happening (and you'll have to take our word for this) is that instantaneously, when L+R is doing the modulating, L–R isn't and vice versa. Actually, each component might have been allowed to modulate the main carrier 100%, were it not for the fact that a bit of 19k Hz signal has to be sent along as well, and it takes up the remaining 10%. The 19k Hz is needed so that at the receiving end it can key the re-introduction of a locally-produced 38k Hz carrier to take the place of the one that was stripped away at the transmitting end.

Because of this refinement, the mono listener notices virtually no degradation of signal quality when listening to a stereo program. Unhappily, however, the FCC's concern with the mono listener was not lavished upon the new or potential stereo convert. Since the L–R information is used to *AM* modulate the sub-carrier, the signal-to-noise ratio of a received stereo program in stereo can never be as good as that of a monophonic FM signal. In fact, it is some 13 decibels (nearly four times) worse. It is for this reason that so much emphasis has lately been placed upon the need for a sensitive tuner or receiver as well as a good, outdoor antenna for successful, noise-free FM stereo reception.

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the technical quality of records and tapes

Reviews are concerned with audio reproduction qualities, not musical performances by Edwin S. Bergamini

Obviously, jazz combos are two or more players. Soundwise, we recently looked into recordings of combos from two to seven players, in different instrumental makeups.

Two players are apt to be piano and string bass. Indeed, the ivories and the doghouse form the base for many another group. With more than two, this pair, or percussion (drums, brushed or struck cymbals, bass drum, etc.) and bass are the driving core of the combo. From there, the group is built with one or more saxophones (there are baritone, tenor, alto, and soprano saxes), trumpet and/or trombone, clarinet, flute (sometimes bass clarinet and alto flute), guitar—and the remarkable "vibes" (vibraphone).

Considering the stereo side of the recording problem, *spread* is most important. Depth isn't a big factor where two to seven guys are (surely if you found them in a night spot somewhere) working out of a space that in some cases could actually fit between your stereo speakers!

How much background? Recording "takes" of jazz groups are either done on location where the boys happen to be appearing at the time (and you get all the noise the crowd happens to be making, too), or they're actual recording sessions, with studio silence prevailing. For this report, we limited our listening to the more precise products of recording sessions. Note, too, that while we like it to be quiet for best listening, quiet doesn't mean a dead, echoless sonic ambiance. No "graveyard studios" for us! We quickly found out, too, that the engineers responsible for all the recordings listed in this tale seemed to feel that way, too.

It turned out that our first listening choice was a mistake. We selected (for what we surmised might well be the smoothest, most faithful sound) a tape as the basis for comparison. This was **"Two of a Mind" (RCA Victor FTP-1172), featuring a group of four: Paul Desmond, left, on alto sax, Gerry Mulligan, right, on baritone** sax, percussion left rear center, string bass right rear center (and closer to the center than the percussion). Good semicircular spread, you might say; and it was indeed. But the saxes, especially the alto sax, tended to be overmiked, sounding just too gritty to be pleasant throughout.

But turning to another **Desmond tape by RCA Victor, "Take Ten" (FTP-1151)**, all was joy. Here was the same spread, with that same fine semicircular feeling to it (guitar, bass, percussion, and sax as you move from left speaker to right)—but this time the saxophone (and everything else) was miked just right, and clean as a whistle. (Interesting thing: t'was the same recording engineer, both times. You can't win 'em all.)

Now before some one suggests we simply don't like a good real gutty sound from sax, drums, or bass, let's get one thing clear quickly. Whatever sound it catches, good recording is a precise and *civilised* process. It no more makes sense to record the Desmond sound too closely (for maximum "water hammer" effect in the pipes, one might say) than it did (on the classical side, some years ago) to record Pablo Casals in a cello concerto so closely that every scrape and scratch (and every grunt and wheeze from the soloist) was mercilessly documented. We're supposed to be listening to music, remember? Those who find the strictly secondary noises of trapped saliva, clicking keys, etc. useful for checking the response of their sound systems might just as practically turn to a sound-effects recording, or one using test tones. They'll get no sympathy from this corner.

So the word is: don't move in too close with that mike! And as it turned out for the rest of this report, nobody did.

"'S Make It," with Art Blakey and the Jazz Messengers (Limelight LS86001), found on the other hand, six players ideally placed and grouped. On the left side, Blakey's percussion-stuff spoke from the (not too far) corner, with rearish piano, and tenor sax forward and toward the center. Trombone (nearer) and trumpet filled out the right channel. And string bass seemed right on the center line between speakers. And sonically, too, everything was right, miked to perfect distance. Both brasses, trumpet and trombone clean, clean. The cymbal in the percussion shimmered and sizzled by but never hissed or spat (how're speakers, good reader, after this one?). In short, nothing was wrong at all—and the idealist in us suggests it could well be this way all the time.

Letters

Send your audio questions, problems, comments and suggestions to the Editor, AUDIOFAN 25 West 45th St. New York, N.Y. 10036

DEAR AUDIOFAN:

I would like to see more technical articles and a complete review of a record—not just a mechanical one.
Edward Stromberg
Jackson Heights, N. Y.

Judging from the many letters we receive applauding AUDIOFAN'S sonic review approach to records and tapes, you're in the minority.—Ed.

DEAR AUDIOFAN:

Will you have a writeup on machines and tapes for automobiles?
Dr. W. M. Greenhut
Scarsdale, N. Y.

January's AUDIOFAN just did. But let's hear more about hi-fi subjects you'd like us to cover. —Ed.

DEAR AUDIOFAN:

In the September issue of AUDIOFAN, your answer in *Ask The Audiofan Answerman* to a question on "volume expanders" has aroused my interest. Saying that "Many of the best recordings now have about as much dynamic range as a lot of listeners want in their living rooms" seems to me to miss the point of what "high fidelity" means. Even before reading your column I had come to the conclusion that no music reproduction system will truly deserve the appellation "high fidelity" until it reproduces the *entire* dynamic range of live music. I dare say that many listeners do not want the same range of sound pressures on their ears at home as they experience at concerts. But many people prefer the abnormal boost of the high bass region.
Craig Stark
Somerset, N. J.

TUNING IN

- The true tone of instruments is often modified on recordings, say some people. Researchers at the University of Illinois apparently agree. The University's Dr. David Freedman uses a computer to break down musical sound waves on recordings into voltage equivalents and, using mathematical equations for each tone produced by instruments, reassembles the tones the way they should sound. It's claimed that the electronic transformation makes the reproductions sound more like the original tones.

- More and more pre-recorded reel-to-reel tapes are being bought, reports indicate. Capitol Records, for example, says they're running 200% ahead of last year's tape sales. The Company's tape sales break down to 75% popular, 25% classical. 3¾ ips is the big speed for this record-tape company who produces 7½ ips pre-recorded tapes only when there is not enough material to fill out a two-reel package.

- A recent advertisement by a manufacturer of mass-produced package phonos, Packard Bell, attributes all sorts of wonderful things to its wood cabinetry, which is called a "high fidelity component." Says the manufacturer, ". . . [the wood cabinetry] adds magnificent resonance to the beautiful sound . . ." Aw, c'mon fellers, wood shouldn't add anything to the sound; hi-fi sound, that is. It's flattering, though, to find the mass producers of home entertainment equipment using audio expressions such as "component," "resonance," and other terms familiar to audiofans.

- Automobile supplier companies are not taking back seats in the auto tape player industry. Orrtonics, one of the three basic tape player contenders, is owned by Champion Spark Plugs. In this light, it doesn't come as a great surprise to learn that the Orrtonics player might be used by Chrysler. (Champion Spark Plugs is a Chrysler supplier.) Lear got there first, however, and Chrysler will follow Ford's lead in having auto tape players as original equipment options in their autos. Ford, which uses Motorola-made auto tape players and Lear continuous loop cartridges, claims that 15 to 20 percent of 1966 Thunderbirds are sold with auto tape players. Lincolns trail with 12 percent, while standard Fords hit 3.5 percent. The B. F. Goodrich Tire and Rubber Co. is jumping into the auto tape player field, too, it has been announced, with both 8-track and 4-track machines.

- The home video tape recorder market is as active as a live volcano. Matsushita Electric Corp. (Panasonic), for example, throws its hat into the HVTR ring by announcing a helical-scan unit will soon be available (it's been shown at trade shows). It uses a system not unlike Sony's, that is, rotating video heads and a stationary audio-control head. Using 12 ips, it eats up somewhat more tape than does Sony's (7½ ips) and Ampex' (9.6 ips). An optional adapter unit is needed to play through conventional TV receivers. Of special interest to audiofans is Panasonic's provision for separate audio erase. Thus, a user can add separate sound effects or insert whatever audio he wishes.

- Another Japanese firm, Akai Electric, is expected to have a video tape recorder here next year. Whereas HVTRs recently introduced use helical scan recording methods, the Akai employs fixed heads. This calls for very high tape speeds, around 90 ips at this time, but a ¼" tape, rather than the ½" or 1" used by helical scan recorders, would lessen the cost of tape somewhat, though it would still be much higher than that of slow speed recorders. Picture quality, too, might leave room for improvement since it's said to be under 200 lines (the U.S. system of television works on a 525 line standard). This fixed head system does offer something extra —it doubles as a stereo 4-track tape recorder.

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